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UNIVERSITI TEKNOLOGI PETRONAS

A SEAMLESS VERTICAL HANDOFF PROTOCOL FOR ENHANCING THE PERFORMANCE OF DATA SERVICES IN INTEGRATED UMTS/WLAN NETWORK

by

SYED SAFDAR ALI RIZVI

The undersigned certify that they have read, and recommend to the Postgraduate Studies Programme for acceptance of this thesis for the fulfillment of the requirements for the degree stated.

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SEPTEMBER 2014

DECLARATION OF THESIS

Title of thesis	A Seamless Vertical Handoff Protocol for Enhancing the Performance of Data Services in Integrated UMTS/WLAN Network

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hereby declare that the thesis is based on my original work except for quotations and citations which have been duly acknowledged. I also declare that it has not been previously or concurrently submitted for any other degree at UTP or other institutions.

Witnessed by

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Date :	Date :

DEDICATION

I dedicate this research work to my loving parents, brothers and sisters.

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All praise to Allah (SWT), who created a man and taught him what he knew not, “Who has taught (the writing) by pen”, and taught man which was not possible without Allah’s guidance. May Allah grant peace and honor to messenger of Islam, Muhammad (PBUH) and his family.

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ABSTRACT

The Next Generation Wireless Network (NGWN) is speculated to be a unified network composed of several existing wireless access networks such as Wireless Local Area Network (WLAN), Global System for Mobile (GSM), Universal Mobile Telecommunications System (UMTS), Worldwide Interoperability for Microwave Access (WiMAX), and satellite network etc. The NGWN will permit the wireless client to roam across diverse integrated access networks while maintaining the session continuity. In order to maintain the session continuity in such an internetworking environment, the most important and challenging issue is to attain the seamless mobility. The literature demonstrated that when the wireless client switches the active data session between the heterogeneous wireless access networks, the loose coupling based mobility management techniques lack in providing the seamless mobility. Conversely, by implementing significant modifications in the existing network architectures and protocols' design, the tight coupling based interworking mechanisms efficiently address the seamless mobility. Nevertheless, these mobility management solutions are not acceptable by the telecommunication/internet service providers since intensive modifications raise the network implementation and operational complexities whilst increasing the network deployment cost. Moreover, usually the contemporary techniques introduce Additional Protocol Overheads (APOs) for the data transportation when the wireless client is located inside the foreign network. These overheads waste the scarce wireless bandwidth and escalate the intermediate nodes data processing time. Consequently, increase the application response time and decrease the network throughput.

This dissertation bridges the gap between seamless mobility and network simplicity. In this regard, a Seamless Vertical Handoff Protocol (SVHOP) is proposed, designed, and evaluated for the integrated UMTS/WLAN network. In order to provide the uninterrupted data connectivity during handoffs, the foundations of the

underlying principle of SVHOP is developed on the proactive link layer Received Signal Strength (RSS) based vertical handoff algorithm along with the optimized multi-homing capabilities of the mobile client. By these features, less signaling and node processing cost influences the overall handoff process. Consequently, low handoff delay and low packet loss during upward and downward vertical handoff cases are achieved. In addition, in this work, an APO free data transportation mechanism has been proposed. The additional protocol overhead free mechanism escalates the packet processing performance of intermediate nodes by using the IP address swapping technique. The synergy of network simplicity and APO free data transportation mechanism lead to achieve fast application response time of various internet applications and increases the network throughput.

ABSTRAK

Terdapat spekulasi mengatakan Jaringan Tanpa Wayar Generasi akan datang (NGWN) akan menjadi rangkaian gabungan dari beberapa rangkaian tanpa wayar yang sedia ada seperti Wireless Local Area Network (WLAN), Global System for Mobile (GSM), Universal Mobile Telecommunications System (UMTS), Worldwide Interoperability for Microwave Access (WiMAX), rangkaian satelit dan sebagainya. NGWN akan membenarkan pelayan tanpa wayar mengunakan pelbagai akses rangkaian bersepadu dalam sesi yang sedang berlangsung. Dalam usaha untuk mengekalkan kesinambungan suatu sesi dalam inter-rangkaian, kelancaran mobiliti adalah isu yang amat dititikberatkan dan paling mencabar. Kajian awal menunjukkan bahawa apabila pelayar tanpa wayar beralih sesi data aktif antara pelbagai rangkaian akses tanpa wayar, pengurusan mobiliti gandingan longgar gagal dalam menyediakan mekanisma mobiliti yang lancar. Sebaliknya, dengan melaksanakan pengubahsuaian yang signifikan ke atas binaan asas rangkaian sedia ada dan reka bentuk protokol, mekanisma gandingan mantap bekerjasama dengan cekap untuk mobiliti yang lancar.Walaubagaimanapun, penyelesaian pengurusan mobiliti yang dicadangkan tidak digunapakai oleh pembekal telekomunikasi/perkhidmatan internet kerana modifikasi yang intensif akan meningkatkan perlaksanaan rangkaian dan kerumitan operasi serta meningkatkan kos penempatan rangkaian. Di samping itu, kebiasaannya protokol kontemporari akan memperkenalkan “Additional Protocol Overheads” (APOs) untuk menghantar data apabila pelayar tanpa-wayar berada di dalam rangkaian asing. APO akan menghadkan jalur lebar tanpa-wayar dan menambahkan titik perantaraan masa pemprosesan data, mengakibatkan peningkatan masa untuk bertindakbalas dan mengurangkan masa untuk pemprosesan rangkaian.

Secara umumnya, kajian ini bertujuan untuk mengkaji penemuan diantara mobiliti lancar dan rangkaian yang mudah. Secara keseluruhannya, “Seamless Vertical Handoff Protocol” (SVHOP) telah dicadangkan, direka dan dinilai untuk rangkaian

UMTS/WLAN bersepadu. Dalam usaha untuk menyediakan sambungan data tanpa ganguan semasa “handoffs”, perkara asas di dalam prinsip SVHOP telah dibina di atas lapisan pautan proaktif “Received Signal Strength” (RSS) berdasarkan algoritma “handoff” seiring dengan kemampuan “multi-homing” yang dioptimumkan untuk pelayar mudah alih. Dengan ciri-ciri ini, kekurangan isyarat dan titik pemprosesan akan mempengaruhi proses “handoff”. Oleh yang demikian, kelewatan “handoff” yang rendah dan “packet loss” sewaktu penaikan dan penurunan menegak “handoff” telah dicapai. Tambahan pula, di dalam kajian ini, mekanisma pegerakan data APO yang bebas telah di cadangkan. Mekanisma APO bebas akan meningkatkan prestasi pemprosesan paket nod pertengahan dengan menggunakan teknik pertukaran alamat IP. Sinergi kesederhanaan rangkaian dan APO bebas akan memacu mekanisme untuk mencapai masa tindak balas yang pantas untuk pelbagai aplikasi internet dan meningkatkan pemprosesan rangkaian.

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LIST OF ABBREVIATION

1G	First Generation
2G	Second Generation
3G	Third Generation
3GPP	3rd Generation Partnership Project
3GPP2	3rd Generation Partnership Project 2
4G	Fourth Generation
AAA	Authorization, Authentication and Accounting
AP	Access Point
APN	Access Point Name
APO	Additional Protocol Overhead
BA	Binding Acknowledge
BSS	Basic Service Set
BU	Binding Update
CL	Convergence Layer
CN	Correspondent Node
CS	Circuit Switched
DAD	Duplicate Address Detection
DMMT	Dual Mode Mobile Terminal
EAF	Ethernet Adaptation Function
EIR	Equipment Identity Register
ETSI	European Telecommunication Standards Institute
FA	Foreign Agent
GGSN	Gateway GPRS Support Node
GIF	Internetworking Gateway Function
GSM	Global System for Mobile
HA	Home Agent
HHO	Horizontal HandOff
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force

IG	Internetworking Gateway
IMSI	International Mobile Subscriber Identity
ITU	International Telecommunication Union
MD	Movement Detection
MIP	Mobile Internet Protocol
MN	Mobile Node
MT	Mobile Terminal
MTC	Mobile Terminal Controller
NA	Neighbor Advertisement
NA	Neighbor Advertisement
NFC	Network Switching Function
NGN	Next Generation Network
NGWN	Next Generation Wireless Network
NS	Neighbor Solicitation
PDN	Packet Data Network
PDP	Packet Data Protocol
PDU	Packet Data Unit
PS	Packet Switched
QoS	Quality of Service
RA	Route Advertisement
RAB	Radio Access Bearer
RAN	Radio Access Network
RAN	Radio Access Network
RD	Route Discovery
RNC	Radio Network Controller
RNCE	RNC Emulator
RO	Route Optimized
RR	Return Routability
RRC	Radio Resource Control
RS	Route Solicitation
RSSI	Received Signal Strength Indication

SCTP	Stream Control Transmission Protocol
SGSN	Serving GPRS Support Node
SGSNE	SGSN Emulator
SIP	Session Initiation Protocol
SVHOP	Seamless Vertical Handoff Protocol
TLLI	Temporary Logical Link Identifier
TR	Traffic Received
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Access Network
VGSN	Virtual GPRS Support Node
VHO	Vertical HandOff
VMAC	Virtual MAC
WAG	WLAN Access Gateway
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Network
WPAN	Wireless Personal Area Network
WWAN	Wireless Wide Area Network

CHAPTER 1

INTRODUCTION

1.1 Integrated Heterogeneous Wireless Networks

At present, there are a number of wireless networks such as Wireless Local Area Network (WLAN), Universal Mobile Telecommunications System (UMTS), Global System for Mobile (GSM), Worldwide Interoperability for Microwave Access (WiMAX), satellite etc., that can provide internet access to mobile users. Every wireless access technology has been designed to provide data services according to the particular needs and demands of the users. Consequently, wireless networks exhibit some inherent strengths and weaknesses in terms of service cost, coverage region, and data rates etc. Users always want to have high bandwidth, low cost, ubiquitous services connectivity irrespective of its location and the network being used. This is practically not possible by using just any single wireless network. To consolidate the best features of every wireless technology, the integration of wireless networks has been widely recommended and studied.

It is envisioned that the Next Generation Wireless Networks (NGWNs) or Fourth Generation (4G) networks will incorporate several Radio Access Networks (RANs) to form an overlaid integrated wireless heterogeneous network. In such an internetworked environment, as illustrated in Figure 1.1, a roaming wireless client can continue its ongoing session to any available network. Therefore, a true anywhere at any time data service can be provided when the user is roaming across the heterogeneous wireless access networks. Moreover, at any location where the wireless client is receiving signals of multiple wireless access networks, the best network for the active data session can be selected according to the client preferences and ongoing

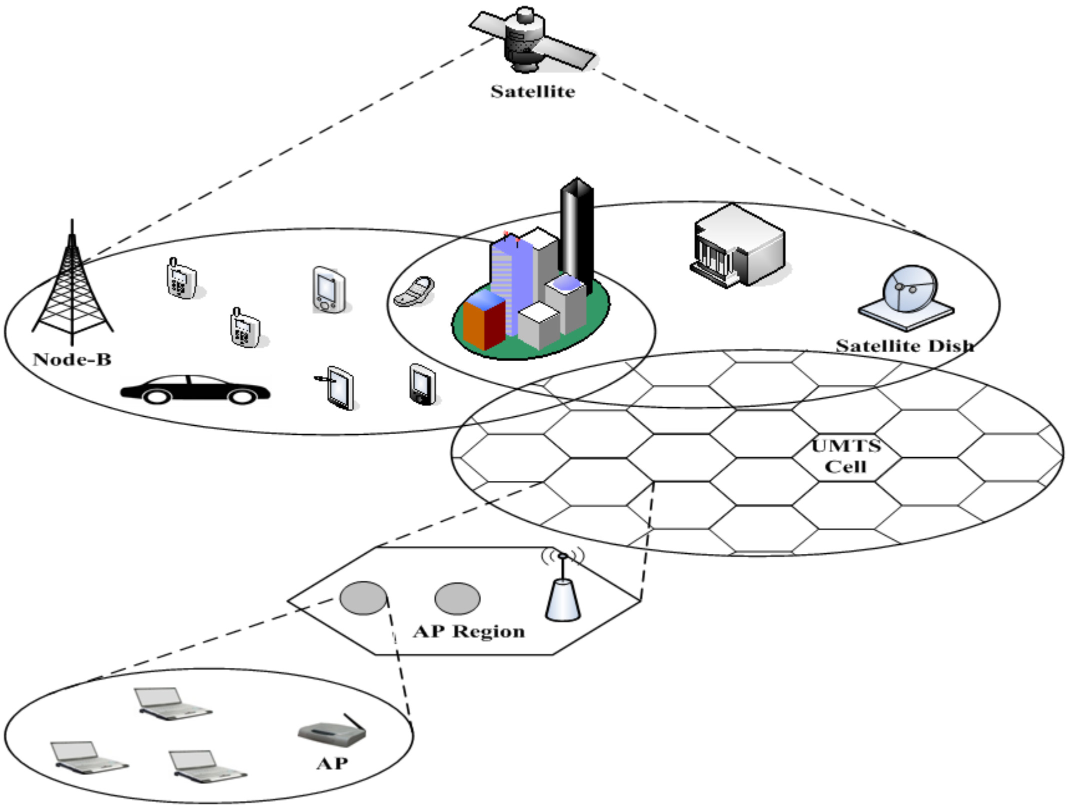

Figure 1.1: Overlaid wireless heterogeneous internetworking system

data session requirements among the variety of available access networks of a converged overlaid network.

Such network integration mechanism or protocol by which the active data session can be switched between heterogeneous wireless access networks is still far to be realized. Up till now, in order to switch the connection from one access technology to another access technology, the wireless device has to stop the active session from the current network and has to establish a new data session with the target access network.

To provide a unified integrated internetworking system of different wireless technologies, enormous research efforts by the wireless technology research community can be found in the literature. The ongoing research trend shows that the two most promising wireless technologies i.e., UMTS and WLAN integration and the analysis of the handoff viability between them attained the highest attention among the researchers compared to other wireless technologies incorporation [1].

The purpose and architecture of UMTS and WLAN networks are completely different. However, the diversity of their features complements each other to establish an integrated network. Consequently, despite a competitor access network, the WLAN network is now considered as a true complementary access technology for the cellular operators.

1.2 Technical Challenges

The UMTS and WLAN are two different technologies that came into existence to serve different goals and fulfill diverse service demands. Therefore, the implemented protocols, algorithms, data rate, authentication mechanisms, handoff mechanism, coverage ranges, etc., of these technologies are dissimilar. Therefore, to design a unified integrated network architecture that consolidates heterogeneous wireless access networks brings a lot of technical challenges and problems to deal with.

For the integrated wireless networks, besides establishing compatibility between two different wireless networks, another critical task is to design a multimode user device. The multimode mobile device must have the following functionalities:

a) It should be able to discover the existence of different access networks.

b) On the basis of either user or network defined policies, the multimode terminal must be able to select the best network among the various wireless access networks for the session continuity.

Once a suitable core network and multimode mobile terminal is designed, the next challenge is to deal with the seamless mobility management. The NGWN has brought a unique challenge, which does not exist before the concept of network integration, known as Vertical Handoff (VHO). The VHO can be defined as the handoff between radio access networks which are representing different technologies, for example, UMTS and WLAN. To ensure the uninterrupted services with minimum Quality of Service (QoS) degradation while the user is roaming across the heterogeneous wireless access networks, the vertical handoff mechanism must be seamless i.e., handoff delay and packet loss must be within the tolerable ranges. As discussed earlier, seamless mobility management between the UMTS and WLAN network has attained the highest attention among the researchers and still require a lot more contributions from the relevant research community.

1.3 Problem Statement and Motivation

In order to realize the true anywhere at any time data services, one of the most important and challenging issues is "seamless vertical handoff" [2]. A handoff is called seamless when it provides both [1, 3, 4]:

1) Fast i.e., low latency handoff and

2) Smooth i.e., no or very little packet loss during handoff.

To ensure the fast vertical handoff, it is important to minimize the signaling traffic [5]. Along with seamless mobility, the ease of network implementation is one of the most prominent parameters that need to be taken into prime consideration [2]. As explained in [6], a best integrated network design must keep the modifications as less as possible in the existing network design.

A comprehensive literature review in the area of integrated heterogeneous wireless networks has revealed that existing mobility management protocols are still lacking in providing a sophisticated seamless vertical handoff mechanism. Existing solutions for the mobility management encompasses Session Initiation Protocol (SIP), Stream Control Transmission Protocol (SCTP), tight coupling and two variations of Mobile Internet Protocol (MIP) i.e., MIPv4 and MIPv6.

It has been observed that MIPv4 mechanism has some serious drawbacks such as:

1. Triangular routing: The triangular routing problem of MIPv4 increases end to end delay which is unacceptable for real time services [7-11].

2. Additional protocol overheads: The IP-in-IP encapsulation introduces 20 bytes overhead to every data packet [12, 13], which slow-down the processing speed of intermediate routers and network devices.

3. Seamless mobility lacking: The long duration of the MIP handoff process significantly increases the high handoff latency and packet loss during handoffs [11, 14, 15]. Consequently, MIPv4 is unable to provide seamless mobility.

4. Additional network components: Introduction of additional mobility management components in the existing networks is another shortcoming of MIPv4 approach [7].

The MIPv6 mechanism addresses many MIPv4 shortcomings such as triangular routing problem. Moreover, provides the lower vertical handoff delay and packet loss compared to MIPv4 mechanism. However, literature [16-18] dictates that the MIPv6 is also not an appropriate choice to attain the seamless mobility. Furthermore, similar to MIPv4, MIPv6 also introduces Additional Protocol Overheads (APO) and required additional mobility management devices.

In order to reduce the handoff delay and packet loss, the tight coupling mechanism has been widely suggested in the literature. The tight coupling mechanism reduces the handoff latency and packet loss during the session handoff between the heterogeneous access network because:

a) Fewer number of signaling messages are required to execute the handoff and

b) In contrast to MIP, intra-domain mobility management is implemented on the integrated network which reduces the signaling transmission cost.

Nevertheless, the main drawback of the tight coupling technique is the high operational complexities compared to any other mobility management protocol. This happens because the tight coupling mechanism highly depends on the underlying wireless access technologies. Furthermore, the internetworking protocols are highly inspired by the existing UMTS mobility management protocols. Therefore, in order to establish the compatibility among different network interfaces, intensive modifications in the existing protocols and network architecture is required [19-21]. In addition, similar to the MIP, the tight coupling mechanism introduces additional mobility management components and uses APO for the data transportation in the integrated network.

When the desired mobility management tasks cannot be accomplished from lower layers, the researchers start finding the solution at the application layer. The SIP is a well-established protocol that has been used for session management in cellular networks. The SIP permits the independent network deployment and traffic engineering, in addition, it does not require modifications in the Correspondent Node (CN). The former function was missing in tight coupling mechanism, whereas, the latter was not provided by the MIPv6 mechanism. Despite many strong features, SIP protocol has its own inherit issues. For example, because of application layer operations, SIP cannot re-establish a broken TCP connection when the IP address gets changed during/after handoff [7, 12, 22]. Therefore, SIP does not provide a transparent terminal mobility [22]. Moreover, for the SIP based mobility management, several SIP servers are needed to be installed in the core network. Various literature articles demonstrate that SIP lacks in providing seamless mobility, since it provides several seconds of handoff latency [23, 24].

The SCTP is one of the latest protocols applied for the mobility management of heterogeneous wireless networks. A lot of problems which were not properly tackled by the other protocols are gracefully addressed by using the SCTP. The most attractive features of SCTP are:

- Only end nodes participate in executing handoffs.
- No modification in the intermediate routers and no additional mobility management components are required to be installed inside the existing networks.
- Handoff latency is just few hundreds of mili-seconds

Nonetheless, the main obstacle in the deployment of the SCTP is the dependency over the SCTP transport services. The SCTP is designed to eventually replace the TCP and UDP transport layer services. Even though, the SCTP is many aspects better than TCP and UDP, however, implementation of SCTP based transportation services are quite less. This is because of the fact that currently almost every device software implementation is based on TCP and UDP. The SCTP selection for the transport service will obviously require reconfiguration of existing devices. The devices over which TCP and UDP protocols are being operated as the transport layer protocols are literally beyond to any counting mechanism. A global replacement of the widely deployed TCP and UDP based devices is extremely difficult, if not impossible [25].

Furthermore, another very important mobility management parameter that is missing when mSCTP solely deal with the mobility management is the “location management” [26, 27]. Since, the SCTP is an end-to-end protocol in which only the end points participates and updated with the Mobile Node (MN) relocation. Consequently, if the MN moves to the foreign network then new session establishment request cannot be facilitated by the home network, since it would be unaware with the MN relocation. Therefore, in order to provide the location management, mSCTP can be used along with the MIP or SIP [27].

It can be observed from the above collected information that most of the contemporary protocols lack in providing the seamless mobility such as MIPv4, MIPv6 and SIP. The tight coupling mechanism can be considered as one of the best protocols that ensure the seamless mobility while the wireless client is moving across integrated wireless access networks. However, because of high modification in the existing data networks, such solutions are not appreciated. Moreover, most of the above mentioned protocols required additional protocol overheads for the data transportation when the wireless client moves to the foreign network. Consequently, increases the intermediate router’s processing delay, and do not efficiently consume the scarce wireless bandwidth.

Therefore, it is required to have an integration protocol that can provide seamless mobility whilst keeping the network modifications as little as possible. Furthermore, to enhance the data service experience of the end user, it must avoid the additional protocol overhead for data transportation in the foreign network.

1.4 Research Hypothesis

The research presented in this thesis is carried out by hypothesizing that the issue of seamless vertical handoff can more effectively be addressed by using the multi-homing technique and proactive Received Signal Strength (RSS) based vertical handoff protocol by decreasing the handoff latency and packet loss during upward and downward vertical handoff cases. It is further hypothesized that the APO free mechanism can enhance the performance of data services by decreasing and increasing the application response time of various internet application and aggregated network throughput, respectively.

1.5 Research Aims and Objectives

The aim of this research is to provide an end-to-end network integration solution to provide anywhere at any time data services to the wireless client without any significant degradation of data services. The performance of data services is analyzed in terms of handoff delay, packet loss, additional protocol overheads, application response time, network throughput etc. This research also aims to study the effects of data network coupling mechanism so that modification in the existing networks can be minimized.

In this perceptive, the primary objectives of this thesis are as follows:

- To develop a protocol that reduces handover latency and packet loss during VHOs.
- To achieve enhanced performance of data services in foreign networks by avoiding APOs.
- To develop a simple mechanism i.e., without any significant modification in the existing networks, for the integration of UMTS and WLAN networks.

1.6 Research Scope and Limitations

The solution proposed in this study offers seamless vertical handoff protocol (SVHOP) for the integration of UMTS/WLAN network. The SVHOP provides seamless mobility while the wireless client is roaming across the UMTS and WLAN network. Afterwards, simulations are carried out to evaluate the performance of the SVHOP in comparison with contemporary protocols. This dissertation includes the study of handoff delay, session blackout time, packet loss, lost information, additional protocol overheads, application response time, and aggregated system throughput. Nevertheless, the study of parameters like blocking probability, connection dropping probability, handoff failure probability, and probability of unnecessary handoff do not fall under the scope of this research work. The proposed mechanism is implemented when the wireless clients are either static or moving with the pedestrian speed. However, the high speed wireless is not considered in this research work. The proposed mechanism utilizes RSS as the link layer hint to execute the vertical handoff. The effects of other parameters like Signal to Noise Ratio (SNR) and current load of the target network on the performance on the vertical handoff decision are also not considered in this thesis.

1.7 Chapter Summary and Thesis Organization

This chapter provides an overview of integrated heterogeneous wireless networks. The prime motivations and benefits of integrated wireless heterogeneous networks are also discussed. Afterwards, this chapter highlighted the gap between the state of the art and desired 4G goals. In order to bridge this gap, the Seamless Vertical handoff Protocol is proposed along with the overall research objectives. The remainder of the thesis is organized as follows.

Chapter 2 can logically be divided into two parts: (1) background studies and (2) literature survey. The background study illustrates the evolution of wireless communication from 1G to 4G networks along with their advantages and shortcomings. Moreover, mobility management in NGWN, its classification, and requirements of handoff management has been discussed. Furthermore, a detail discussion on the basic components of UMTS network with their important features and mobility management protocols are demonstrated. In addition, an overview of WLAN networks has been presented.

The literature survey portion of this chapter summarizes the seminal contributions of state-of-the-art from various standard organization bodies and the research community. The taxonomy of heterogeneous wireless network design and internetworking protocols are also overviewed. Finally, this chapter presents a qualitative analysis of existing mobility management protocols which clearly highlights the need of further elevation in the field of wireless heterogeneous networks.

Chapter 3 discusses the design considerations, network topological design (ranging from the core network to the wireless client), implementation and evaluation of the proposed SVHOP. This chapter demonstrates how the proposed mechanism addresses several issues that were not properly managed by the existing protocols. A detailed upward and downward vertical handoff performance analysis of the benchmarks and proposed mechanism are also been presented in this chapter.

Chapter 4 presents the simulation network design, parameters, traffic, scenarios etc., used for the implementation and evaluation of proposed SVHOP. In addition, the proposed mechanism is compared to the contemporary internetworking protocols.

Chapter 5 concludes the undertaking research work with the future work recommendations.

Lastly, the list of research publications related to this research work is provided.

CHAPTER 2

BACKGROUND KNOWLEDGE AND LITERATURE REVIEW

2.1 Chapter Overview

This chapter provides the overall background knowledge and related work relevant to this research study. The background study portion of this chapter starts by discussing the evolution of wireless communication from 1G to 4G networks along with every particular generation's advantages and shortcomings. Afterwards, mobility management in Next Generation Wireless Network (NGWN) with its classification and handoff management requirements has been explained. Subsequently, the most important concept of NGN i.e., vertical handoff has been discussed. In order to understand the related concepts of UMTS and WLAN network, the sections come afterwards demonstrate the basic components of the UMTS network along with their basic functions and mobility management protocols. The background part of this chapter is completed by presenting the WLAN network and its different topological designs.

The literature survey portion of this chapter summarizes the seminal contributions of state-of-the-art from various standard organization bodies and the research community. The taxonomy of heterogeneous wireless network architecture and internetworking protocols are also overviewed. In order to illustrate the motivation of this research work, a detailed discussion on several internetworking contemporary protocols which correspond with different TCP/IP layers has been discussed comprehensively. Finally, this chapter presents a qualitative analysis of existing mobility management protocols which clearly highlights the need of further advancements in the field of wireless heterogeneous networks.

2.2 State of the Art

The following subsections reflect the state of the art in the field of integrated wireless heterogeneous networks. Several standardization bodies have been contributing in the area of integrated wireless heterogeneous networking, such as 3rd Generation Partnership Project (3GPP), Institute of Electrical and Electronics Engineers (IEEE), Internet Engineering Task Force (IETF), International Telecommunication Union (ITU), 3rd Generation Partnership Project 2 (3GPP2), and European Telecommunication Standards Institute (ETSI) etc. Nevertheless, the most widely accepted and appreciated suggestions are obtained from the 3GPP, ETSI, and the IETF. The 3GPP has defined six scenarios that demonstrate the high-level internetworking requirements (e.g., network charging, authentication, selection etc.) [28]. The 3GPP internetworking scenarios are discussed in the next subsection. The ETSI has defined two generic approaches for the integration of UMTS and WLAN; namely: loose coupling and tight coupling, which is overviewed in Section 2.2.2. In addition with the 3GPP and ETSI, the IETF has defined several network integration protocols at different TCP/IP layers which are discussed in Section 2.3.

2.2.1 3GPP WLAN Interworking Standardization

One of the primary standardization bodies that developed the 3G mobile cellular systems was the 3GPP. Most of it efforts were focused on addressing the convergence of data and voice communications. In addition, one of the stated goals of 3GPP was the attainment of up to 2Mbps data rate in the indoor environment [29].

Nevertheless, the 3GPP has identified that the mobile systems are still lacking in providing the quality coverage inside the buildings. Moreover, it has been recognized that the integration of UMTS and WLAN networks can bring several advantages to the wireless client. For example, the wireless client can be facilitated with higher bandwidth and quality coverage of WLANs inside the buildings. In addition, service continuity of the ongoing session can be attained with the help of the UMTS network.

Table 2.1: Summary of the 3GPP internetworking scenarios

	Scenario 1	Scenario 2	Scenario 3	Scenario 4	Scenario 5	Scenario 6
Common Billing	X	X	X	X	X	X
Common Customer Care	X	X	X	X	X	X
3G based Access Control		X	X	X	X	X
3G based Access Charging		X	X	X	X	X
Access to 3G PS based Services			X	X	X	X
Access to 3G PS based Services with Service Continuity				X	X	X
Access to 3G PS based Services with Seamless Service Continuity					X	X
Access to 3G CS based Services with Seamless Mobility						X

In order to integrate the advantages of UMTS/WLAN networks, the 3GPP Technical Specification Group Services and System Aspects has published a feasibility study in its TR 22.934 [30], which discussed six internetworking scenarios.

Table 2.1 [30], presents the different UMTS and WLAN internetworking approaches. These internetworking approaches range from the connectivity of two networks only by the means of common billing and customer care and gradually progress to a complete seamless integration of two systems.

- Scenario 1 – Common billing and Customer care: This scenario represents the simplest form of internetworking of two systems. In this scenario, only the common billing and customer care services are provided to the wireless client.

- Scenario 2 – 3G-based Access Control and Charging: This scenario suggests an integration mechanism in which the 3G subscriber can initiate their session through WLAN access networks. The Authorization, Authentication and Accounting (AAA) servers can be used for authentication and charging purposes. Inside the WLAN access network, the 3G subscriber's USIM information can be used for authentication. The basic motivation of this scenario is to provide the IP connectivity or access to internet with the help of WLAN access networks.

- Scenario 3 – Access to 3G Packet Switched services: In this scenario, the 3GPP suggests an internetworking scenario in which the WLAN client can access the packet based services of the 3G network. For example, Multimedia Messaging Service (MMS) and Wireless Application Protocol (WAP) services can be utilized from the WLAN network. To gain the access to the 3G network, the data can be tunneled from 3G core network to the WLAN access point.

- Scenario 4 – Access to 3G Packet Switched-based services with service continuity: In addition to the access to the 3G packet based services as described in scenario 3, this scenario also includes the data session continuity between the WLAN and 3G networks. For example, if the 3G user initiates its data session from 3G network, it can continue this session even after roaming to the WLAN network and vice versa.

- Scenario 5 – Access to 3G Packet Switched-based services with seamless service continuity: This scenario is one step ahead to the scenario 4. The main goal of this scenario is to provide the seamless data continuity between the integrated networks i.e., aspects like data loss and connection break time during the network switching should be minimized and remain unnoticeable to the wireless client.

- Scenario 6 – Access to 3G Circuit Switched-based services with seamless mobility: In this scenario, the 3G circuit switched services such as voice call can be accessed seamlessly from WLAN network.

These scenarios play a major role in defining the way of coupling or the integration technique. Broadly, coupling schemes are divided into two classes: loose coupling and tight coupling which are discussed in the next subsection.

2.2.2 Integrated UMTS/WLAN Network Architectures

In the literature, several types of integration schemes have been discussed [31-34]. Depending on the network inter-dependence, the European Telecommunication Standards Institute (ETSI) has defined two generic approaches for the integration of UMTS and WLAN; namely: loose coupling and tight coupling [35]. These two schemes differ in terms of the connecting point of WLAN with the UMTS network, as illustrated in Figure 2.1. The loose coupling suggests an internetworking scenario in which WLAN and UMTS networks are deployed independently. Here, the WLAN is connected to the internet; therefore, it maintains indirect connectivity to the UMTS network. On the contrary, the tight coupling indicates that the WLAN is directly connected to the UMTS core network, i.e., Serving GPRS Support Node (SGSN) or Gateway GPRS Support Node (GGSN). In such an internetworking scenario, WLAN appears as another access network of the UMTS core network whilst signaling and data traffic traverse through UMTS network.

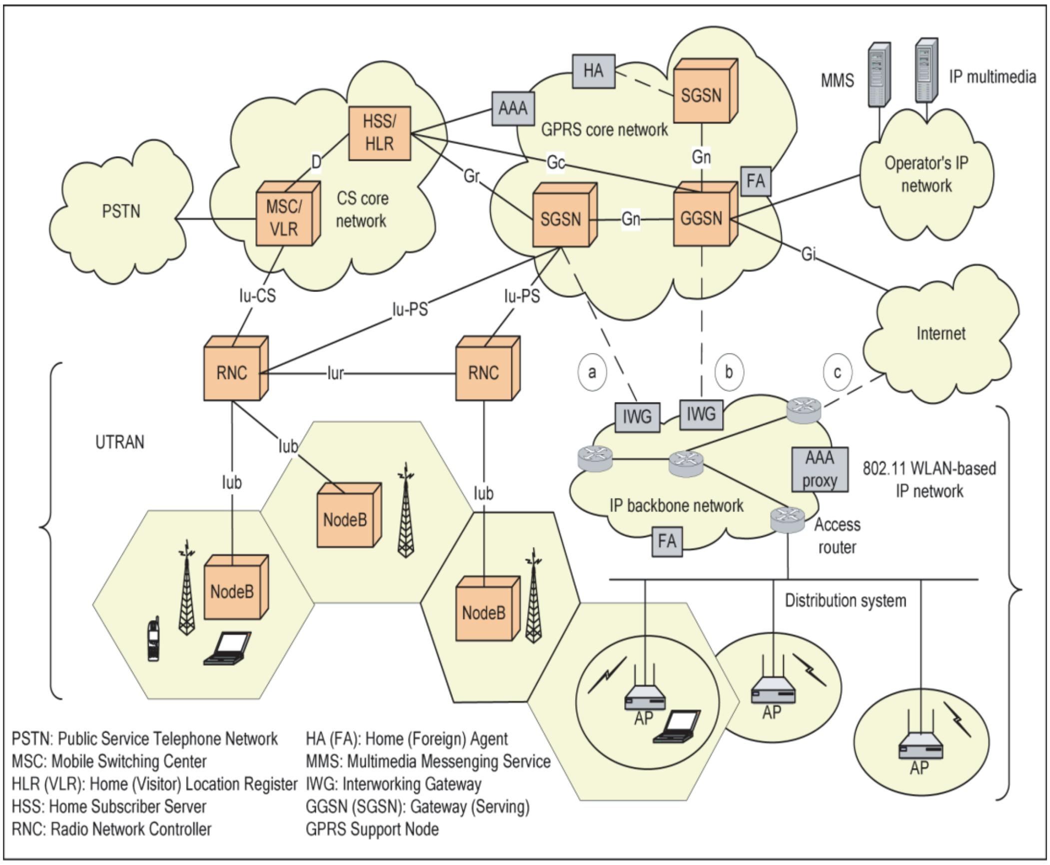

Figure 2.1: Internetworking architecture for UMTS and WLANs: (a) WLAN integrated at SGSN; (b) WLAN integrated at GGSN and (c) WLAN integrated to external IP networks [28]

2.2.2.1 Loose Coupling

In the loose coupling internetworking, networks are deployed and interconnected with each other independently [36]. As shown in the Figure 2.1, from the UMTS network point of view, this interconnecting point exists after the Gateway GPRS Support Node (GGSN), i.e., at Gi interface [35, 37]. Therefore, WLAN network bypasses the UMTS core network for the establishment of a direct connection with the external PDNs and at the same time maintains an indirect connection with the UMTS network. It should be noted that, since the WLAN network is deployed as a complementary UMTS access network and no direct connection between UMTS and

WLAN exist, therefore, the WLAN data does not pass through the UMTS core network rather traverse directly to the internet.

2.2.2.1.1 Pros and Cons

This section briefly discusses the benefits and shortcomings of the loose coupling architecture.

- The main advantage of using the loose coupling scheme is that it allows the independent deployment, traffic engineering and network operations [38]. Consequently, loose coupling integration provides independence to the underlying wireless access technologies [9].

- The high level of network independence permits the network service providers to take advantage of other providers' existing networks [38].

- The main disadvantage of this approach is that the two networks are integrated via the internet. Therefore, signal traffic needs to traverse the long path which causes high handoff latency of hundreds of milliseconds to several seconds [20]. As a consequence of the high vertical handoff delay, high number of packet loss is observed during handoffs.

- For the mobility management, most of the loose coupling approaches require the additional mobility management components. For example, in case of MIP and SIP, Home Agent/Foreign Agent (HA/FA) and SIP servers are required to be installed in the existing networks to perform the vertical handoffs, respectively. This adds the additional node processing delays during handoffs, moreover, increases the total network cost.

2.2.2.2 Tight Coupling

As illustrated in the Figure 2.2, in the tight coupling internetworking approach, internetworking access networks are connected directly to the UMTS core network

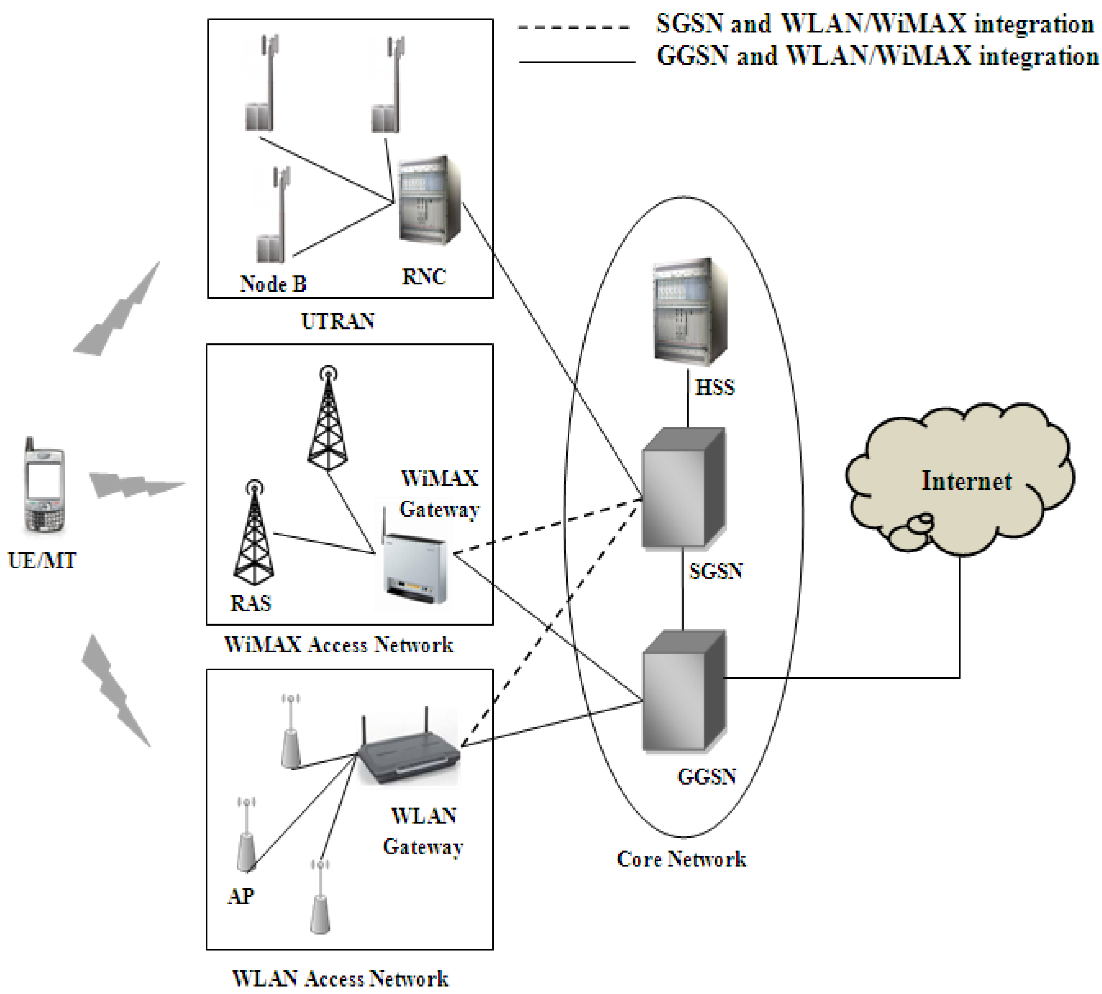

Figure 2.2: Tight coupling points in UMTS network [38]

i.e., UMTS SGSN or UMTS GGSN. This can be achieved by connecting the WLAN/WiMAX network either with the UMTS Iu-PS, Gb [35] or Gn interface. In such an internetworking scenario, the gateways are connected to the UMTS network in the similar manner as it is another UMTS Radio Access Network (RAN). The gateways are introduced to provide compatibility between the two dissimilar networks. In the case of UMTS/WLAN internetworking, the WLAN gateway hides the details of the WLAN network, in addition, implements the functionalities of the specific UMTS network component [39]. Consequently, the UMTS network deals with WLAN as it is its own RAN and finds no difference between UMTS and WLAN access networks. The WLAN data traffic traverse to the internet via UMTS network, therefore, UMTS features such as mobility management, security, authentication, etc., can be applied to WLAN networks. For internetworking, each network needs to modify its services, protocols and interfaces.

2.2.2.2.1 Pros and Cons

This section briefly discusses the benefits and shortcomings of the tight coupling architecture.

- The main advantage of the tight coupling approach is its efficient mobility management mechanism. This efficiency is realized because the two networks are coupled tightly. Consequently, by minimizing the handoff delay and packet loss during vertical handoffs, an intra-domain equivalent handoff performance can be achieved.

- Furthermore, UMTS core network resources, subscriber database, billing system and authentication mechanism can be reused. This reuse of resources leads to low cost network deployment [40].

- The main drawback of the tight coupling technique is the high operational complexities compared with the loose coupling technique. This happens because the existing mobility management protocols of the UMTS network are applied over the integrated network. Therefore, in order to establish the compatibility among different networks intensive modifications in the existing protocols and network architecture is required [19-21]. Moreover, as depicted in Figure 2.2 for the protocols conversion and data routing between UMTS and WLAN networks, additional network emulators/gateway devices are required [9]. Since, the internet service providers/ telecommunication operators all over the world have already invested an enormous amount to the fully functional legacy networks; significant modifications in existing protocols and the introduction of additional mobility management components would not be easily accepted.

- Tight coupling integration solutions are highly dependent on the integrating wireless access technologies [9]. Therefore, if more and more networks are integrated together, more gateways/emulators would be required for the protocol conversions.

- Furthermore, as the data traffic of WLAN traverse via the UMTS network, they potentially create a bottleneck in the UMTS network. Therefore, in order to avoid this, modifications in the existing UMTS nodes are required to sustain high traffic loads [21].

2.3 Mobility Management Protocols in Heterogeneous Wireless Networks

As illustrated in Figure 2.3, several integration protocols under different coupling approaches have been proposed by the IETF and different research bodies at different layers of TCP/IP protocol stack for the integration of heterogeneous wireless networks. Every mobility management protocol has its own advantages and shortcomings. This section is divided into four sub-sections. Each sub-section discusses the state of the art in the field of heterogeneous wireless integrated networks corresponding to different TCP/IP layers.

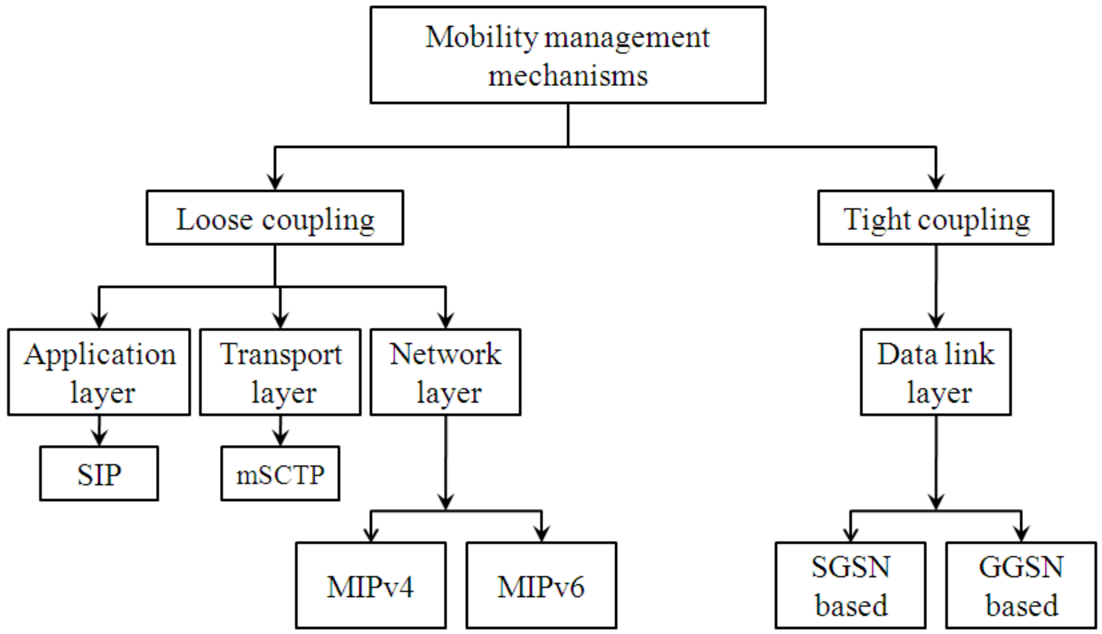

Figure 2.3: Taxonomy of the integration protocols at different TCP/IP layers

2.3.1 Network Layer Mobility Management

For the network layer mobility management, MIPv4 and MIPv6 are the IETF proposed standards. The Mobile IP is the most prominent and widely studied protocol for the mobility management of heterogeneous wireless networks compared to all other mobility management approaches [41, 42]. In order to alleviate the mobility management issues enormous contributions have been made by the researchers by implementing MIPv4 and MIPv6 in heterogeneous wireless internetworking environment. The MIP4 and MIPv6 are discussed in following subsections.

2.3.1.1 MIPv4

In data communication, every device is uniquely identified by its IP address. The IP addresses are not only used to identify the node, moreover, they are used to route the data packets. In case of static source and destination, both devices are required to have a single IP address for their unique identification and data routing. However, for example, if the destination is wireless mobile client which is moving between different access networks along with an ongoing data session with the internet server (source) then whenever it leaves its home network the ongoing data session will be broken. This happens because of the absence of the destination device at its home network.

In order to provide the session continuity while the MN is roaming between different access networks, the MIPv4 [43] has been proposed. This article has been recognized as the Landmark article of the previous decade. In order to maintain the session continuity of the mobile user, the author suggested that two IP addresses should be used by the mobile client. One is the fixed home address (HoA) and the second is the care-of-address (CoA). The CoA address is assigned to the mobile client at each new point of attachment. Therefore, whenever the client moves from its home network to the foreign network, instead of its HoA, the CoA would be used to route the packets. Nevertheless, the upper layers will be using the HoA. That is how the session continuity along with the upper layer transparency can be achieved. It should

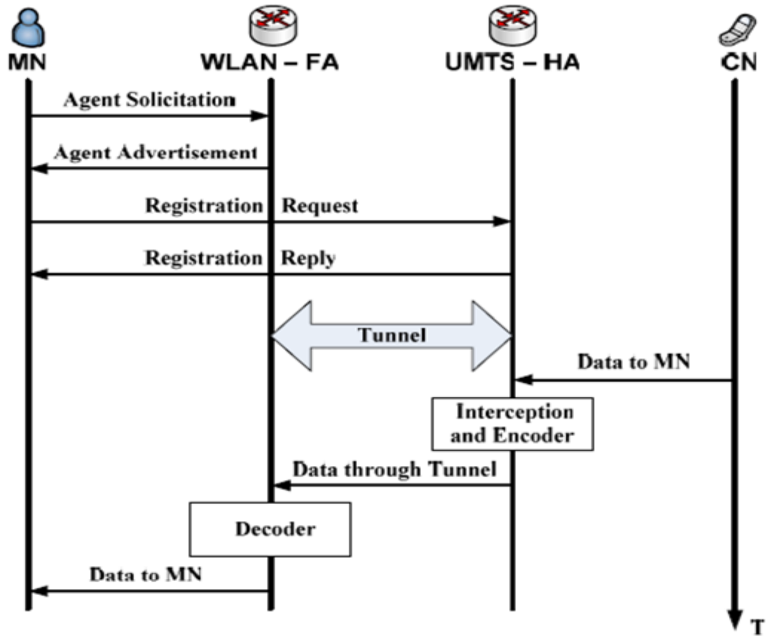

Figure 2.4: Mobility management by using MIP [15]

be noted that, to perform the Mobile IP operations, Home Agent (HA) and Foreign Agent (FA) are required to be installed in the home and foreign network, respectively.

In the literature several research articles [7-15, 22, 26, 44-49] have elaborated MIP implementation, its advantages and shortcomings. Functionally, the MIP can be divided into three basic operations [9, 47] i.e., agent discovery, registration, and tunneling.

When the MN is located within its home network, the traditional routing mechanism is applied. However, when the MN moves from its home network to the foreign network, as illustrated in Figure 2.4 [15], it discovers the presence of the foreign network via Agent Advertisements (AAs) broadcasted by the FA. With the help of the AAs, the MN learns the CoA. After the agent discovery procedure and CoA learning, the MN sends a Registration Request to the Home Agent (HA). By using the Registration Request message the MN informs the HA about its relocation.

As illustrated in Figure 2.5 [15], since, the Correspondent Node (CN) is unaware of the MN relocation, therefore, it keeps on sending the data to the MN home

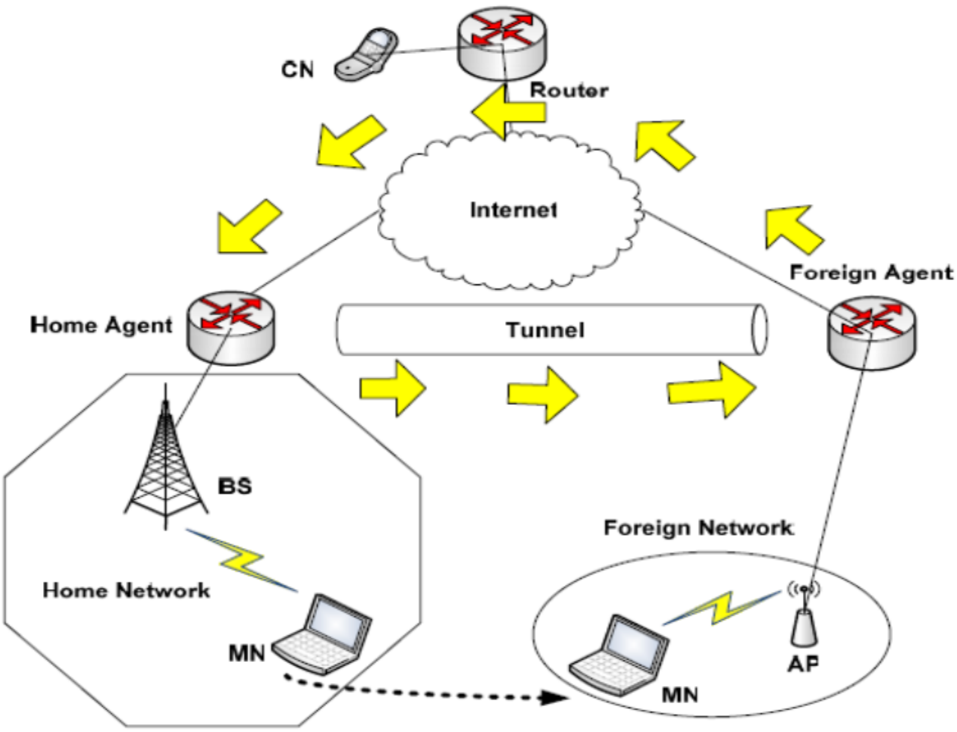

Figure 2.5: Mobile IP triangular data routing and tunneling [15]

network. The HA retransmit the packets to the MN associated FA by establishing an IP-in-IP tunnel. This encapsulation is used to alter the normal IP routing of packets. The HA and FA act as the end point of the tunnel. The HA sends the encapsulated packets and the FA is responsible to decapsulate them and send the original IP packets to the MN. Finally, when the data are receipted at MN, in the packet destination field the MN finds it home address. Consequently, the upper layers remain transparent to the MN relocation. For the data from the MN to CN, the foreign network directly sends packets to the CN. As shown in Figure 2.5, this asymmetric way of data routing from and to the MN is called as triangular routing.

Despite providing the session continuity, unfortunately the MIP mechanism has some serious drawbacks.

- Triangular Routing: As illustrated in Figure 2.5, when the MN is moved to the foreign network then in order to communicate, the datagrams are first routed from CN to HA then HA send the data packets to MN. However, for the data packets transfer from MN to CN the datagrams can be forwarded directly from

MN to CN. The additional routing introduced by this triangular routing mechanism highly increases end to end delay compared to optimal routing which is not acceptable for the delay sensitive applications like VoIP, etc. [7, 9-11, 21, 22]. The authors in [12] and [11] reported that in case of datagrams transmission from CN to MN within a campus network, the MIP increases end to end delay by 45%. Moreover, the end to end delay is expected to be increased when the two entities are physically far apart (for example, in case of WAN).

- Additional Protocol Overheads: When the MN is moved to the foreign network then the HA tunnels the data packets to the MN's new point of attachment. Introduction of additional encapsulation overheads while establishing the data tunnel is one of the major drawbacks of MIP [7, 8]. The IP-in-IP encapsulation significantly adds the overheads which is particularly very critical for the real time services [22]. The encapsulation is performed in a manner that the original datagram is wrapped with the outer IP envelope. Consequently, the final packet contains the outer IP header as well as the inner datagram. Typically, 20 bytes overhead is appended to every data packet [12, 13], which simply waste the network available bandwidth. Furthermore, slow down the processing speed of intermediate routers and network devices [13]. Therefore, in addition to the triangular routing, additional encapsulation overheads are another factor that further increases the network latency.

- Seamless Mobility Lacking: Seamless mobility is the prime objective when designing any mobility management protocol for the integrated network design. It has been reported by several studies [11, 14, 15] that the long duration of the MIP handoff process significantly increases the high handoff latency and packet loss during handoffs. Consequently, MIP is unable to provide seamless mobility.

- Additional Network Components: As discussed earlier, the MIP mechanism requires to install the mobility agents i.e., HA and FA in the home and foreign networks, respectively. Introduction of mobility agents is considered as the shortcoming of MIP approach [7], since, it increases the cost of network deployment.

2.3.1.2 Mobile IPv6

Due to the several shortcomings of MIPv4 as discussed earlier, it was desired to introduce an efficient protocol that can overcome the MIPv4 limitations. For this, MIPv6 has been launched. The MIPv6 operates in two modes: bidirectional tunneling and route optimization.

The bidirectional tunneling can be considered as the enhanced extension of MIPv4. In this mode of communication, the packets from the CN to the MN are first routed to the HA and then tunneled to the MN. However, from MN to CN data transmission the packets are first tunneled to the HA and then routed normally to the CN. Here, the IPv6-to-IPv6 encapsulation is used [44]. Similar to the MIPv4, in the MIPv6 bidirectional tunneling mechanism the CN does not need to be updated with the MN relocation. Therefore, here is no need to install the MIP feature’s implementation on the CN side. In addition, the FA deployment is not required in the MIPv6, for both bidirectional and route optimization cases, because of MIPv6 stateless auto configuration feature. Therefore, the network deployment is simpler than MIPv4.

However, similar to the MIPv4, the bidirectional tunneling mechanism also suffers with the high additional encapsulation overheads and end to end delay problems. Since, all data are traversing through the HA, therefore, HA can be a single point of failure. In addition, in case of increasing mobile devices HA can become the network performance bottleneck. These limitations bring the researchers' attention towards the MIPv6 route optimization mechanism.

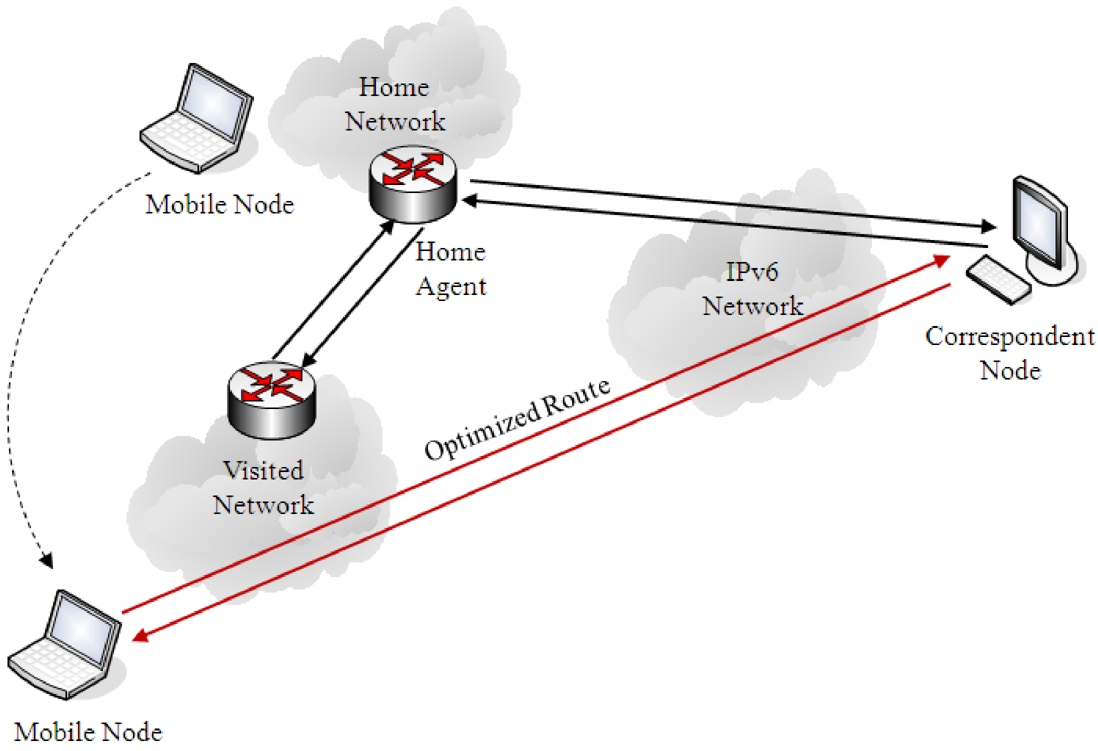

Figure 2.6: MIPv6 route optimization

Despite many similarities with the MIPv4, the major advantage of the MIPv6 route optimization mechanism is the avoidance of several shortcomings of MIPv4 and bidirectional tunneling mode. For example, FA deployment is not required. Moreover, the IP-in-IP encapsulation and triangular routing are avoided by introducing the route optimization technique. As illustrated in Figure 2.6, in contrast to the MIPv4 mechanism and bidirectional tunneling mechanisms in which the packets are routed from the HA, the MIPv6 route optimization allows the shortest path communication between the mobile devices by enabling the CN and the MN to send the data packets directly to each other. Consequently, the congestion in the home network and HA can be eliminated.

To establish a direct communication path, the MN not only sends the registration request to the HA, moreover, the CN is also updated with the MN relocation. Since, the CN is aware with the current CoA of the MN after relocation, therefore, instead of sending data packets to the HA, the CN establishes a direct communication path with the MN. This direct communication mechanism helps to reduce the end to end delay.

In case of integrated UMTS/WLAN network, the handoff latency of a roaming MN can be measured as the time when MN stop sending/receiving packets from the previous access network to the time when the MN starts sending/receiving data packets to the new access network. In an integrated UMTS/WLAN network, the Mobile IPv6 vertical handoff operations are performed as illustrated in Figure 2.7. It should be noted that the signaling procedure of MIPv6 depicted in the Figure 2.7 is inspired by the MIPv6 mechanism implementation presented in [4, 50].

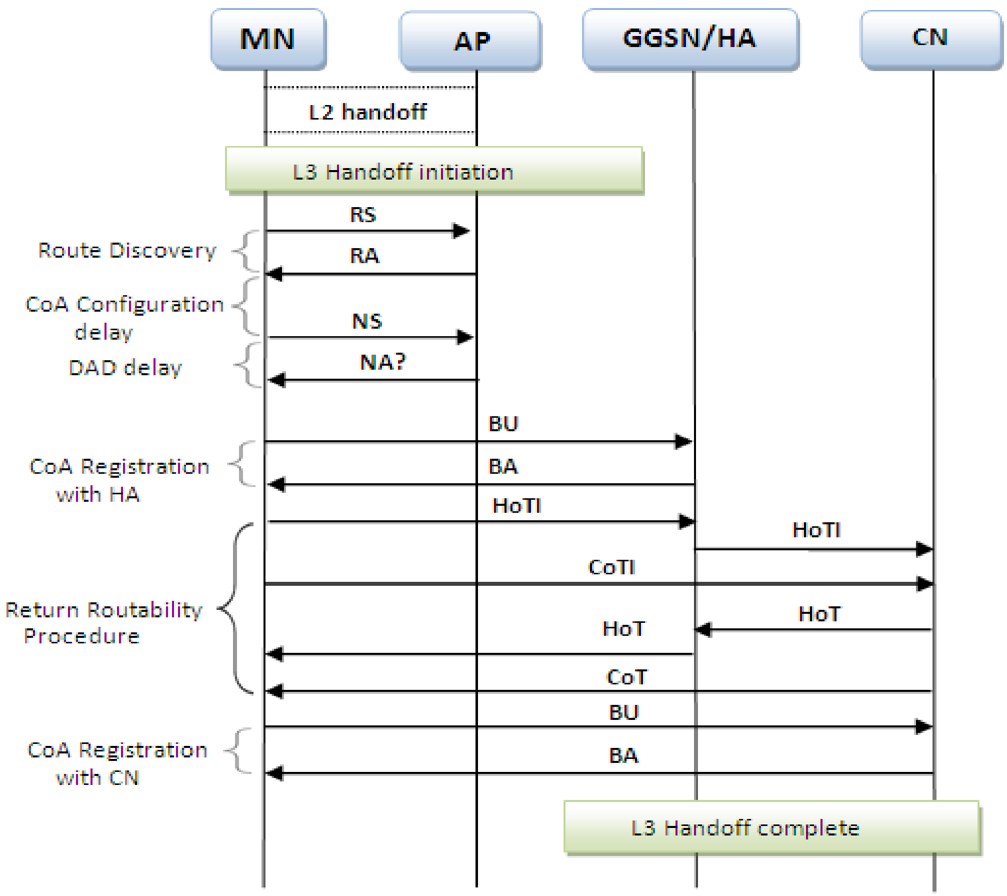

Figure 2.7: MIPv6 signaling during handoff in an integrated UMTS/WLAN network

When the MN moves and detects the foreign network, it performs Route Discovery/ Movement Detection (RD/MD) mechanism. The RD contains the handshake of Route Solicitation (RS) message sent by the MN, which is responded back with the Route Advertisement (RA) message of the foreign network access point. In order to configure the CoA address, the MN extracts the foreign network prefix information from RA. Based on the network prefix information and MAC address of MN, a CoA is generated by the MN.

According to the IPv6 specifications, the Duplicate Address Detection (DAD) procedure is required to perform to ensure the uniqueness of the configured CoA. The DAD scheme is performed by the MN by sending the Neighbor Solicitation (NS) to the new network and wait for the Neighbor Advertisement (NA) message. If no NA is received in response of NS for the time of 1000ms, then according to the IPv6 recommendation, the configured CoA is considered unique [44]. After the successful unique CoA configuration, the CoA registration procedure is performed by the MN with the HA and CN. To perform the registration procedure to update the relocation information of MN, Binding Update (BU) messages are sent by the MN to HA and CN which are acknowledged by Binding Acknowledge (BA) messages. It is worth remaindering that the Return Routability (RR) is performed between the HA and CN update procedure. Afterwards, all the data packets are routed to the MN at its new location.

Due to the difference in traffic load, ongoing data services on the MN, hardware implementation, access network, and implementation versions the value of the handoff latency highly varies. Therefore, it is difficult to assume a single value of handoff latency for different MIPv6 implementations [50].

In [16], for the integrated UMTS/WLAN network the authors claim that when the UE moves from UMTS to WLAN access network then it is the UE choice whether or not to perform a handoff. Moreover, in this type of handoff significant handoff delay does not appear. However, in case of WLAN to UMTS handoff, the UE suddenly losses the connection with the WLAN without any prior warning. Therefore, after the connection is lost with the WLAN network, the connection establishment procedure with the UMTS network is started. In such handoffs, higher chances of the abrupt ongoing session lost may appear because of the long handoff delay. In order to analysis the WLAN to UMTS handoff delay, MIPv6 has been implemented on the overlay UMTS/WLAN testbed. When the UE lost the WLAN signals, it first establishes the link layer connection with the UMTS network by using the PMM (Packet Mobility Management) protocol and SM (Session Management) protocols.

After that the MIPv6 operations are carried out which are composed of MD, CoA acquisition, and BU. The VHO delay in case of WLAN to UMTS network switching was found to be in between 2 to 4 seconds with the average of 2.863 seconds. Therefore, it has been concluded that the ongoing data session continuity is obtained by the proposed mechanism. However, the seamless mobility has not been achieved which is the ultimate goal of the future networks.

In [51], for the evaluation of MIPv6 handoff a single MN to and fro motion between the home and the foreign network has been considered. The measurements are obtained for two different cases. Namely: optimized and realistic case. In the former case, only single moving MN is considered. However, the latter case includes additional four static active users in the network. It has been observed that for the optimized case, the total MIPv6 handoff latency is around 2.54 seconds. However, when more active users are located inside the network the collisions are more frequent. As a result, time to access the channel will be longer. This increase of L2 handoff latency in the realistic case strongly contributed in L3 handoff latency. Hence, the total MIPv6 handoff latency increases to several seconds.

Sun et al. in [17] have established a test bed which includes WLAN, GPRS and TD-SCDMA. This setup used MIPV6 for the mobility management among the aforementioned wireless technologies. The performance of the defined test bed has been evaluated in terms of handoff delays. The MN performs handoff while downloading the file from the network FTP server. It was observed that the average handoff delay is 1.3 sec, when the MN performs handoff from WLAN to GPRS network. However, the average handoff delay decreases to the average of 0.91 seconds, when MN performs handoff from the WLAN to the TD-SCDMA network.

In [4] and [52] the authors discuss in detail the MIPv6 fundamental steps required to perform the MIPv6 handoff mechanism. As illustrated in Figure 2.8, the MIPv6 mechanism is composed of Movement Detection, Router Discovery, Address Configuration, Duplicate Address Detection, Authentication and Authorization, Register New CoA and Binding update. The authors claim that router discovery and duplicate address detection are the major factors that are too inefficient to attain the seamless mobility.

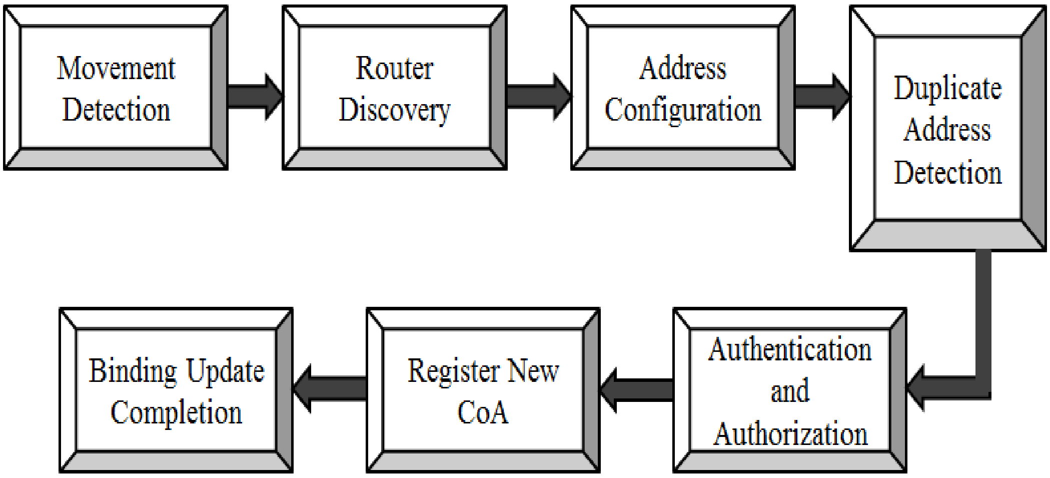

Figure 2.8: Block diagram of MIPv6 handoff steps [52]

In order to analysis the effects of RA on the route discovery process and, hence, on the MIPv6 handoff latency several RA values have been set on the testbed. The results show that the handoff delay is 3.013, 2.448 and 1.917 seconds for the RA interval of 3000, 1000 and 300ms, respectively. It is worth mentioning here, according to the MIPv6 specification [53] the RA interval cannot be set below than 300ms. Although, at the minimum RA interval the MIPv6 handoff latency is decreased significantly, however, it is still too high to attain the seamless mobility. It is concluded that to further decrease the handoff latency the DAD process should be optimized as well.

Similarly, in [54] the authors also identified the DAD delay as the main cause of the handoff latency of MIPv6. They stated that 70% of the MIPv6 handoff delay is contributed by the DAD process. Therefore, in pursuance of reducing the overall MIPv6 handoff latency, a Parallel DAD (PDAD) model was proposed. This model modifies several standard MIPv6 handoff procedures. For example, in contrast to the MN configured CoA, the neighbor routers are responsible to configure the CoA. Moreover, unlike the reactive MIPv6 mechanism, here the CoA address is configured in advance and in parallel by all the reachable neighboring routers. As illustrated in Figure 2.9, when the MN is relocated from its home network it sends the solicitation to all the neighboring routers in parallel. On receipt of RS, all the neighboring router configure the CoA and every individual router checks the uniqueness of configured

CoA in the same network and send the generated CoA to the MN. The MN stores all the received CoA in its cache. From now onwards, if the MN moves to the new subnet it can directly use the stored CoA without performing DAD because the uniqueness of CoA has been already checked by the neighboring routers.

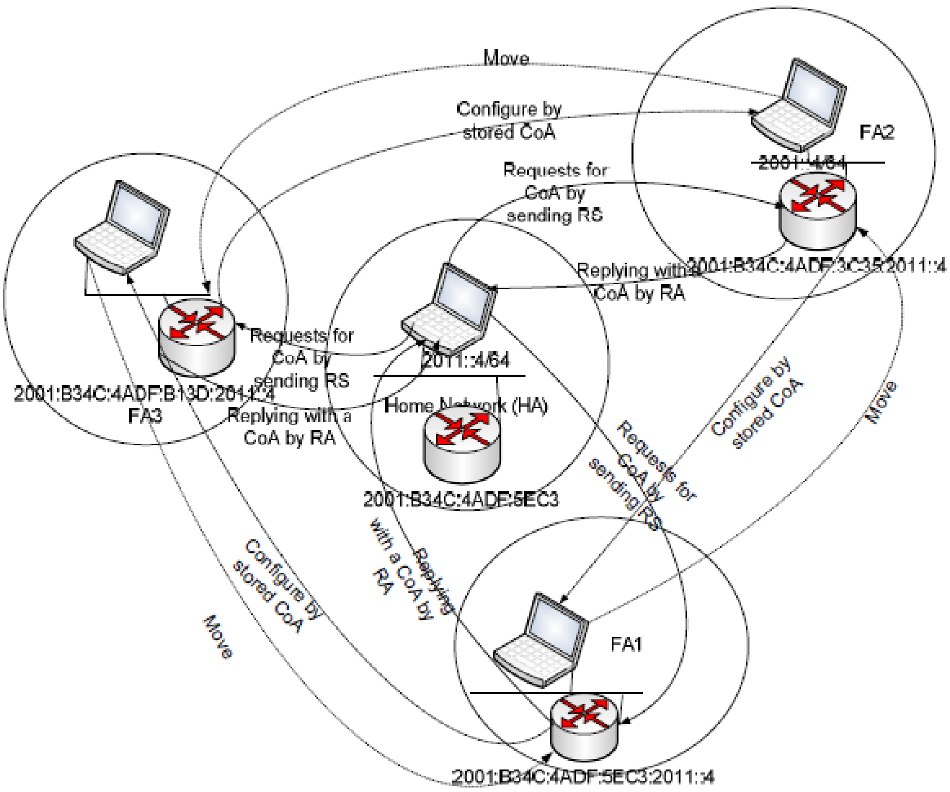

Figure 2.9: Parallel DAD mechanism [54]

The proposed PDAD mechanism somehow looks an elegant mechanism to eliminate the latency produced by the standard DAD procedure. Nevertheless, no analytical or simulation analysis was presented. Moreover, here are some drawbacks of this mechanism. For example, sending requests to all the neighboring routers and collecting CoA from every individual router will be an inefficient utilization of the scarce wireless bandwidth specially in case of big networks. Furthermore, the authors did not mention how long the MN and routers have to store the configured CoAs into the cache. Since, keeping the CoA means unique IP addresses cannot be used by any other device. Moreover, keeping CoA for a long time in the cache will eventually affect the performance of MN and routers.

In [18] the authors discussed the basic framework of MIPv6 in detail. Similar to the [4] and [54], the authors explored that the major source of delay in the MIPv6 mechanism is DAD process. Therefore, in order to improve the network performance in terms of handoff latency, an improved framework for MIPv6 has been proposed. In the proposed mechanism, it is suggested that the DAD process should be replaced by the New Access Router (NAR) configured CoA mechanism. For CoA address configuration, the routers contain two IP address pools: Free address pool and Assigned address pool. Free address pool contains the free IP addresses that can be used to assign the IP address to the new connecting nodes. However, the assigned address pool contains the IP addresses that have already been assigned to the nodes.

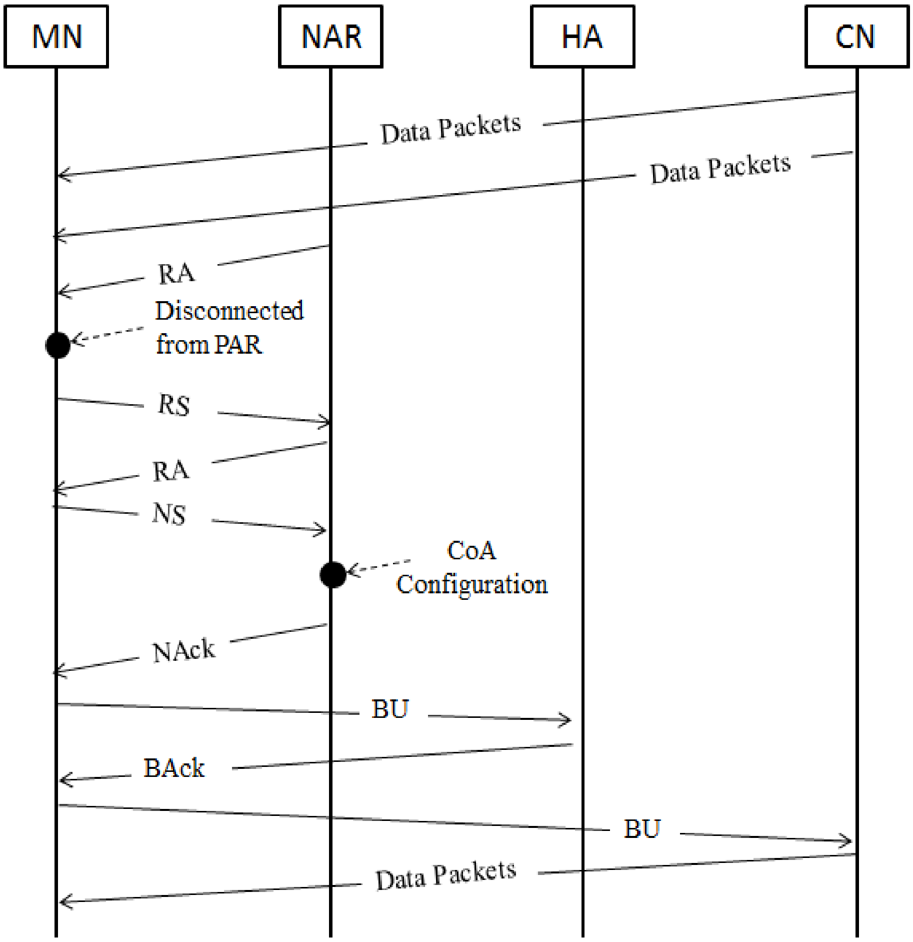

Figure 2.10: Improved Framework signaling procedure during handoff [18]

Figure 2.10 illustrates the signaling flow of the improved framework for MIPv6 during the handoff. It should be noted that the entire handoff procedure of the improved framework of MIPv6 is quite similar to the standard MIPv6 mechanism. The only difference comes from the way of assigning CoA to the MN. As shown in Figure 2.10, after the relocation of MN, it sends the Neighbor Solicitation (NS) message to the NAR. On receipt of NS, the NAR configures the unique CoA from the free address pool. The configured CoA is sent by the NAR to the MN by using the Neighbor Advertisement (NA).

Figure 2.11 demonstrates the timing diagram of the improved framework of MIPv6. The total handoff latency by using the improved framework of MIPv6 (T_{HO_I}) was expressed as [18]:

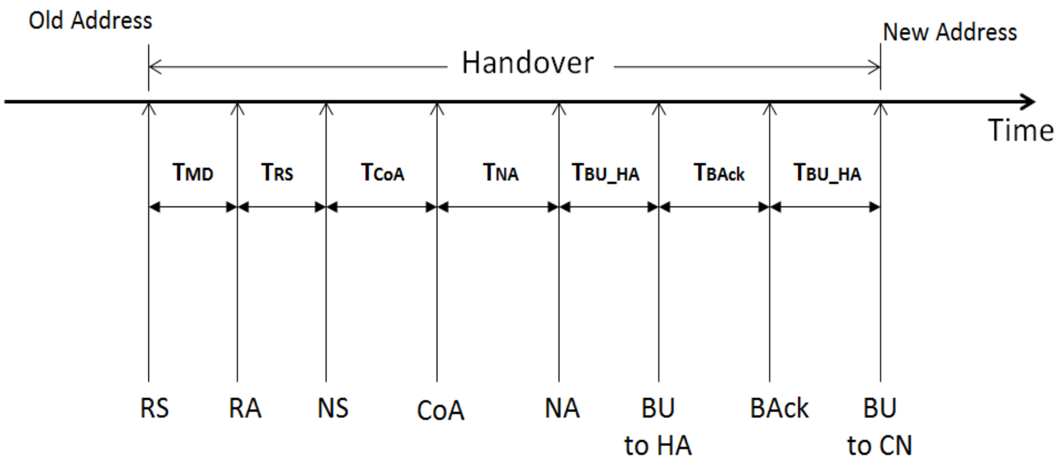

Figure 2.11: Handover Latency in Improved Framework for MIPv6 [18]

$$T_{HO\_I} = T_{MD} + T_{RS} + T_{CoA} + T_{NA} + T_{BU\_HA} + T_{BAck} + T_{BU\_CN} \tag{2.1}$$

Since, the DAD process has been omitted from the proposed mechanism, therefore, it is believed that the handoff latency can be reduced by using the proposed mechanism. In [18]**,** by comparing the handoff latency of improved framework and standard MIPv6 mechanism, the following equation has been presented.

$$T_{Handover\_IMIPv6} < T_{Handover\_MIPv6} \tag{2.2}$$

From the numerical analysis and comparison, the authors concluded that by eliminating the DAD delay of 1.245 seconds from the legacy MIPv6 framework, the handoff delay of 2.943 seconds can be obtained. Therefore, by using the improved framework of MIPv6, the handoff latency is decrease around 30% compared to the standard MIPv6 mechanism.

Vassiliou and Zinonos in [50] implemented a testbed and applied the MIPv6 technique for the mobility management in case of roaming MN across the networks. They identified that several component delays contributed in the total MIPv6 handoff delay. Broadly, the total MIPv6 handoff latency is equal to the sum of L2 and L3 handoff latencies. The handoff latency for the MIPv6 was analytically presented as [50]:

$$D_{MIPv6} = D_{L2} + D_{RD} + D_{DAD} + D_{REG} \tag{2.3}$$

Where, D_{L2}: is the link layer establishment delay; D_{RD} represents movement detection delay; D_{DAD} presents the Duplicate Address Detection and D_{REG} shows the BU/Registration delay. As illustrated in Figure 2.7, the layer 3 delay was further be broken down as [50]:

$$D_{MIPv6} = D_{L2} + (T_{RS} + T_{RA}) + D_{DAD} + (T_{HA\text{-}BU} + T_{HA\text{-}BA} + 2T_{HoTI} + 2T_{HoT} + T_{CN\text{-}BU} + T_{CN\text{-}BA}) \tag{2.4}$$

The experimental setup was analyzed by using default and optimized MIPv6 parameters. For the default parameters case, the DAD delay, RA interval and Router Solicitation delay is 1, 0.5 – 1.5 and 1 second, respectively. With the default values of different Mobile IPv6 latency components the handoff latency of 3.68 seconds was observed. The authors argued that if the MIPv6 mechanism is applied in a controlled environment then the DAD process can be discarded because of the less probability of address duplication in the network. Moreover, the RA interval was set at the minimum value that is 300ms, moreover, the MN was forced to send the RS message as soon as it moved to the new network. Finally, it was shown empirically that the handoff latency can be decreased up to 1.37 seconds by using the suggested optimizing parameter values.

In [55] the authors have analyzed the MIPv6 packet routing efficiency. It is reported that the efficiency of packet routing is one of the most important and yet least studied part of the MIPv6 protocol which is necessary to be fully comprehended before implementing it into the future networks. Both bidirectional tunneling and route optimization mode of communications are analyzed and compared. For the evaluation of application response time, which was considered the prime performance metric, the video conferencing traffic was generated inside the network. It was observed that the application response time highly influenced with the adopted routing mechanism. As expected, when route optimization mechanism was implemented the application responses faster compared with the bidirectional tunneling mechanism. The reasons of the rapid responses were:

a) Less number of route optimization encapsulation overheads (24 bytes) compared with the bidirectional tunneling mechanism (40 bytes). Since, additional encapsulation overheads increase the processing time of intermediate routers and network nodes, therefore, it significantly increases the network latency.

b) The longer transmission cost of the bidirectional tunneling mechanism compared with the route optimization technique. This further contributed in the high end to end delay and, hence, slows down the application response time.

Similarly, in [56], authors emphasis that a better routing approach must avoid the extensive use of APOs. They further added that the issue of overhead is critical, especially in the case of bandwidth intensive applications, because the wireless bandwidth is a scarce resource. Therefore, in order to increase the overall system performance, a tunneling based route optimization technique has been presented. In a standard route optimization case, when the mobile terminal moves to a foreign network with the ongoing session then to make the routing transparent to the upper layers and to avoid ingress filtering, the MN and CN uses home address option and type 2 routing header, respectively. The home address option with the destination extension header and type 2 routing header incurs 24 additional bytes each into the data packets for the communication. However, when the CN is also mobile and relocated to the foreign network then the overall encapsulation overhead will be 48 bytes long.

Similar to the bidirectional tunneling mechanism, the tunneling based route optimization technique uses the standard MIPv6's IPv6-in-IPv6 encapsulation in which the original datagram is wrapped with the outer IP envelope of 40 bytes. However, in contrast with the standard bidirectional tunneling mechanism, the proposed mechanism maintains the tunnel between the MN and CN. Therefore, every CN need to be upgraded. In order to compare the performance of tunneling based route optimization technique and standard route optimization technique, in [56], the overhead ratio has been evaluated by using the following equation.

$$Overhead\ Ratio\ = \left(\frac{Mobility\_Additional\_Size}{Original\_Packet\_Size}\right) \tag{2.5}$$

Based on the findings authors stated that the tunneling route optimization technique includes less additional encapsulation overheads compared to the standard route optimization technique.

In [57], the authors argued that the network layer mobility management solutions like MIP and MIPv6 cannot provide the seamless mobility. In this paper, it has been suggested that the IEEE 802.21 framework, which is also known as the Media Independent Handover (MIH) can be a potential solution to attain the seamless mobility. The MIH places a logical entity between the layer 2 and layer 3 for the seamless mobility realization. In order to execute the vertical handoff, three fundamental steps are required i.e., handover negotiation, handover invitation, and handover execution. Although, the MIH handover mechanism has been elegantly discussed in this research article, however, no simulation results are provided.

The authors in [58] present another MIH based vertical handoff mechanism. For the evaluation of the MIH three different networks are integrated. Namely, UMTS, WLAN, and WiMAX. The load distribution at each network was 5, 10, 15, 20 and 25 nodes with load balancing and without load balancing scenarios. It has been observed that when load balancing algorithm is applied different user exhibits different handover latency values. Nevertheless, in general, it has been found that when the network load is minimum and the load balancing is applied, most of the user suffers approximately 1.2 second handoff delay. In order to guarantee the seamless mobility while the user is moving from one access network to another access network, such high vertical handoff latency is not acceptable.

2.3.2 Link Layer Mobility Management

The link layer mobility management falls under the category of tight coupling mechanism. Section 2.2.2.2 and 2.2.2.2.1 demonstrated the basic tight coupling network design assumptions and its pros and cons, respectively. This section briefly discusses several research contributions based on the tight coupling mechanism. The literature dictates that, as illustrated in Figure 2.12, the tight coupling is attained with the help of either SGSN or RNC emulators. The SGSNE and RNCE are connected with the UMTS GGSN and SGSN, respectively. Therefore, this section is further divided into two subdivisions that separately demonstrate the GGSN and SGSN based integration mechanisms.

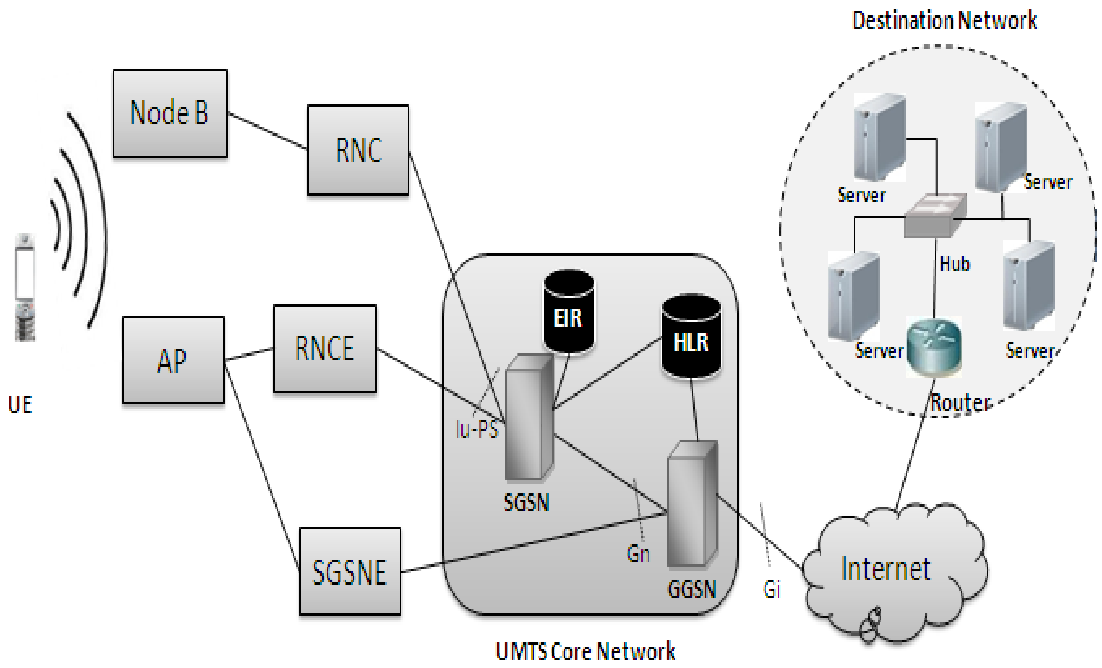

Figure 2.12: SGSN and GGSN based integration network design

2.3.2.1 GGSN based Integration

The GGSN based integration mechanism has been widely suggested and studied by several researchers in the literature [21, 59-63]. When GGSN is the integration point, SGSNE or gateway is required to perform the internetworking. In this type of integration, Gn interface is the integration point of two networks [37, 63]. The SGSN emulator/gateway hides the WLAN access network from UMTS network and performs the functionalities as it were another UMTS SGSN [21].

In [63], the authors overviewed several UMTS and WLAN integration mechanisms that include loose coupling, peer network and tight coupling. In order to understand the tight coupling mechanism, the SGSN emulator approach has been considered. In this coupling technique, the SGSN emulator (SGSN`) has been connected with the UMTS network at Gn interface. The standard UMTS SGSN and SGSN` are identified on the basis of different Routing Area Identity (RAI). Therefore, when the MN roams in between the UMTS and WLAN network, similar to the UMTS inter-SGSN handoff, RA updates are required to be sent to the UMTS core network.

The IP addresses to the UMTS and WLAN network are assigned from the same GGSN IP address pool. Therefore, the mobility of MNs across the UMTS and WLAN networks does not require the IP address change. Whenever the Mobile Station (MS) performs handoff from UMTS to the WLAN network to attain higher bandwidth services, an inter-SGSN RA update is sent. By this, the standard SGSN becomes the old SGSN, whereas, the SGSN` becomes the new SGSN. The GGSN then establishes a tunnel between it and the SGSN`. Now, the packets from the external network will be routed to the GGSN, which in turn send towards the new SGSN i.e., SGSN` and finally the SGSN` will forward the packets to the MS. According to the authors, the peer network architecture is the least complex design compared to other network integration technique because only an AAA server is required to add in the existing network for the mobility management. However, loose and tight coupling designs are much more complex. The given reasons are the requirement of the new additional network components and introduction of the new protocol in the network for the mobility management.

Another GGSN based integration mechanism has been proposed in [62]. In this research contribution, an Internetworking Gateway (IG) has been introduced. The IG performs the conversion of signals and performs the data routing between the UMTS and WiMAX access networks. The main purpose of the designed network was to reduce the handoff latency and packet loss while minimizing the significant modifications in the existing protocol stacks. Similar to other link layer internetworking techniques, during and after the handoff the MN keeps the same IP address. Therefore, no network layer protocols such as MIP are required. The suggested gateway technique has been evaluated and compared with the MIP and another tight coupling technique. It has been observed that the handoff latency of the IG technique is much lower than the MIP mechanism. The reason of faster handoff is that the IG mechanism keeps the same IP address during and after the handoff, however, the MIP IP registration process during handoff adds significant delay in the handoff process. It has also been noted that the less packets are lost when IG technique is applied compared with the other tight coupling technique. This happened because the tight coupling technique burdens the UMTS SGSN and GGSN network for the data routing. However, in case of IG technique the data is passed to the external network via passing only through the GGSN.

In [59], it has been argued that despite many advantages such as the independent network deployment and operations, the major disadvantage of the loose coupling is the long handoff latency and packet loss. They further explain that in the tight coupling network design the WLAN network is directly connected to the UMTS core network. Therefore, because of such close cooperation of the two different networks the WLAN gateways implement all the UMTS radio access network protocols. Since, the tight coupling network solutions are highly influenced with the UMTS protocols, therefore, extensive interface modifications are required in the existing network. Moreover, in order to sustain the increased data traffic from the WLAN access network, the UMTS nodes such as SGSN and GGSN are needed to be upgraded. Nevertheless, due to the intra-domain message exchange during the handoff, the tight coupling provides smooth and fast vertical handoffs. Furthermore, the security, mobility, and QoS of the UMTS network can be reused directly over the WLAN network. On the basis of the above reasons, authors emphasis on the implementation of tight coupling design for the integration of UMTS and WLAN networks compared with the loose coupling architecture. In this paper, a GGSN based UMTS and WLAN integration mechanism has been presented. However, no analytical or simulation results were presented.

Authors of [59] have analyzed their GGSN based integration protocol in [60]. The evaluated protocol is termed as Virtual MAC (VMAC). The VMAC provides a virtual MAC address to the wireless clients when they move from UMTS to WLAN access network. The suggested network design is shown in Figure 2.13 [60]. In order to assign the virtual MAC address to the roaming wireless devices, Gateway Hotspot Support Node (GHSN) has been designed and implemented.

The main functions of the GHSN include:

- UMTS and WLAN radio interfacing.

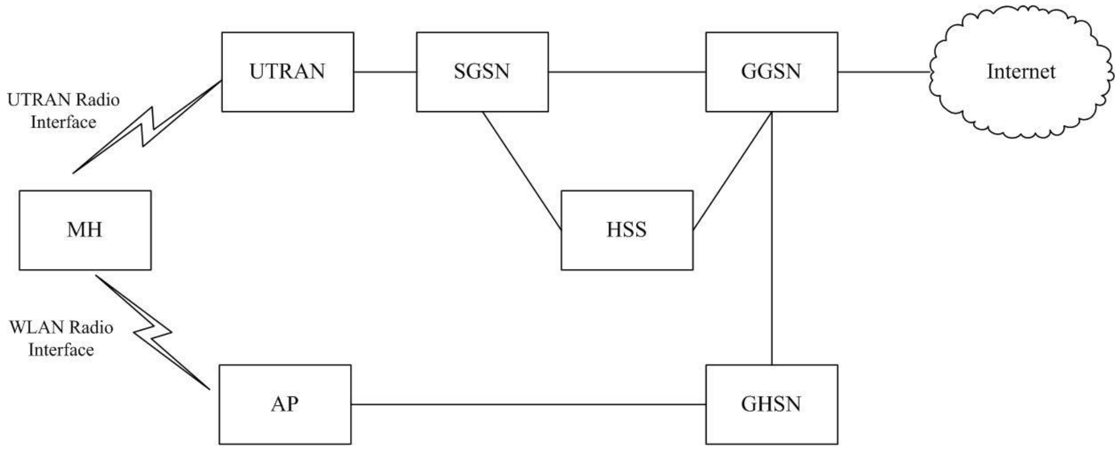

Figure 2.13: Integrated UMTS/WLAN network design [60]

- Based on Mobile Subscriber Identity (IMSI), creation of virtual MAC addresses.

- Forwarding packets between integrated networks.

When the MN moves from UMTS to the WLAN network, it sends a request message to the GHSN for the assignment of virtual MAC address. On receipt of the request, the GHSN configures a VMAC address based on the extracted IMSI from the request message. After that, the GHSN requests the HSS to check the uniqueness of the configured VMAC. When the uniqueness is checked, the HSS stores this address and send an acknowledgement to the GHSN. The GHSN finally responds the MNs VMAC configuration request via WLAN access network. At this stage, the MN is allowed to continue its data session from WLAN network. On the basis of simulation results, it was observed that the handoff latency is 182ms, which was found faster than several compared SCTP configurations. However, only UMTS to WLAN handoff scenario was presented and no simulation results for the WLAN to UMTS network handoff was evaluated.

For the GGSN based tight coupling integration of UMTS and WLAN networks, a new logical node termed as Virtual GPRS Support Node (VGSN) has been proposed in [61]. This node emulates the UMTS SGSN in UMTS network. However, act as an

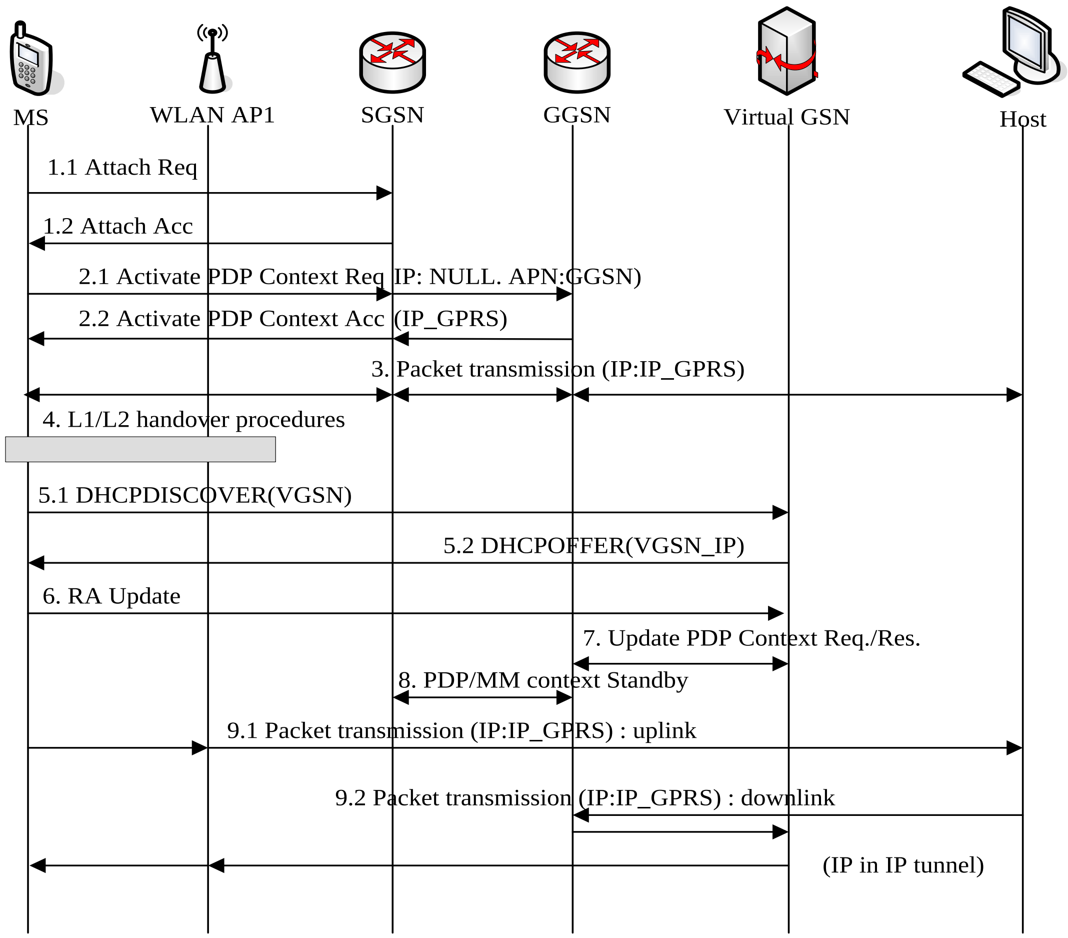

Figure 2.14: VGSN: UMTS to WLAN roaming [61]

access router for the WLAN network. The VGSN routes data packets and converts the control signals between the UMTS and WLAN network for the roaming wireless devices. Since, the two networks are tightly coupled, therefore, the roaming user does not require to acquire the new IP address when roaming between the integrated networks. Figure 2.14 illustrates the signaling method used when the Mobile Station (MS) relocates from UMTS to WLAN. From step 1 to 3 are the standard UMTS procedures which include the GMM attach and PDP context activation procedures. However, when the MS wants to connect to the WLAN network, at step 4, it establishes the link layer connectivity to the WLAN network. After this, based on the Dynamic Host Configuration Protocol (DHCP) the MS finds the IP address of VGSN. After learning the VGSN address, the MS sends it's Routing Area update (RA update)

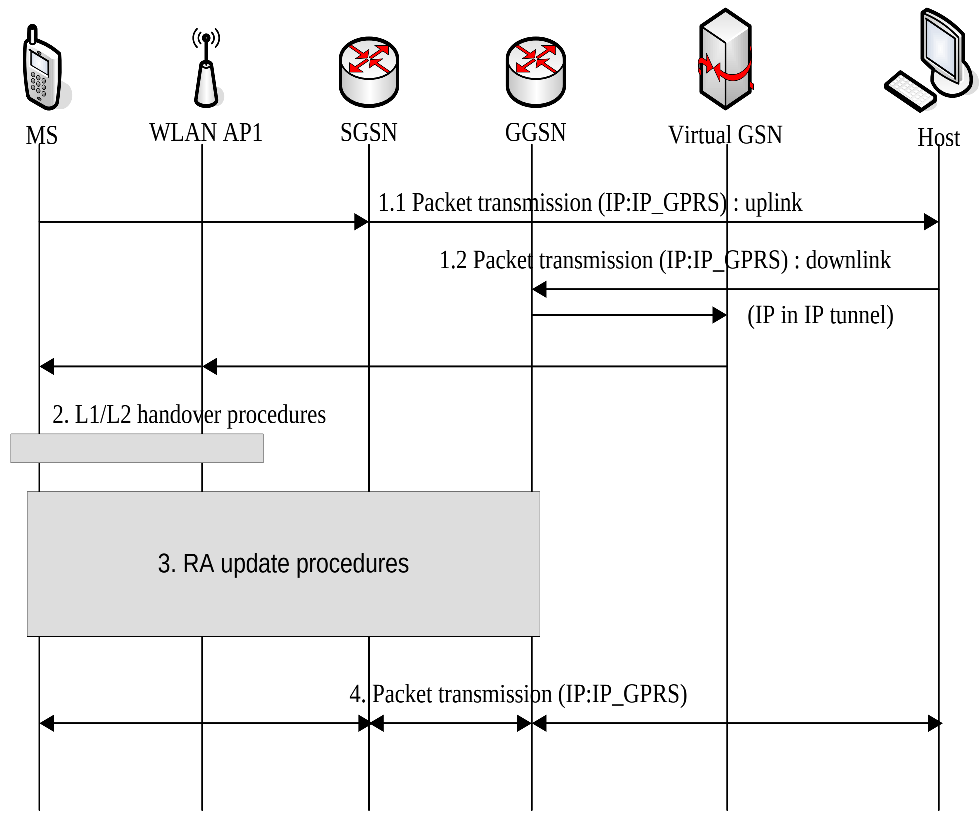

Figure 2.15: VGSN: WLAN to UMTS roaming [61]

to the VGSN by using its UMTS IP address. On receipt, the VGSN sends a standard PDP context activation message to the GGSN and request to change the stored SGSN address with the VGSN address so that the data can now be traversed through it. In order to send the data packets to the MS new point of attachment, the GGSN has to create a tunnel between it and VGSN. Therefore, instead of the old SGSN, the GGSN start sending the packets to the new SGSN (VGSN) which is finally forwarded to the MS via WLAN access network. In order to reduce tunneling overheads, the authors suggested using the IP-in-IP tunnel compared with the standard GPRS tunneling protocol (GTP) tunneling.

The Figure 2.15 shows the signaling mechanism required when the MS moves back to the previous SGSN. In this case, the MS has only need to update the RA information so that the GGSN start sending the data packets to the SGSN instead of the VGSN. The VGSN mechanism has been compared with the MIP. In case of MIP, the signaling cost is higher as the control signal has to pass through the internet. In contrast with this, in case of VGSN the signaling cost is lower as the messages are exchanged within the same domain. Therefore, on the basis of simulation results, the authors concluded that the handoff latency can be significantly decreased by the proposed mechanism compared with the MIP.

In [64], the author proposed a novel network integration design based on IP Multimedia Subsystem (IMS). In this network design, the WLAN is directly connected with the GGSN via SGSN emulator. This model enables the internetworking between WLAN, CDMA 2000, and UMTS networks. For the unified session control, IMS is used at the application layer, whereas, the MIP is used for the IP mobility management at the network layer. For the case of UMTS to WLAN vertical handoff, it has been found that the vertical handoff delay by using the unified MIP and SIP based proposed mechanism was 190ms. On the other hand, the pure SIP based approach produces 302ms delay during vertical handoff.

In order to further evaluate the proposed unified MIP and SIP based IMS network design presented in [64], in [65] the authors integrated UMTS and WiMAX networks. It has been observed that the average vertical handoff latency in case of WiMAX to UMTS and UMTS to WiMAX network handoff are 94ms and 180ms, respectively. In general, it can be deduced that the proposed unified MIP and SIP based IMS network design is a potential candidate to provide seamless mobility. However, the main drawback of this approach is the introduction of additional network components in the existing network design. The additional mobility management components are SGSN emulator, SIP servers, and MIP HA and FAs.

2.3.2.2 SGSN based Integration

For the internetworking of heterogeneous wireless networks, when the integration point is UMTS SGSN, an RNC emulator is required to perform the internetworking between UMTS and WLAN networks. This can be achieved by connecting the WLAN network either with the UMTS Iu-PS or Gb interface [35]. In such an internetworking scenario, the gateways are connected to the UMTS network in the similar manner as it is another UMTS Radio Access Network (RAN). The literature accepts plethora of this type of integration mechanisms by the research community. However, this section summarizes the seminal contributions of the SGSN based integration mechanism.

In order to integrate the WLAN and GPRS networks, a SGSN based integration mechanism has been proposed in [66]. Since, the connected networks are different and tight coupling is used, therefore, for the interoperability between aforementioned

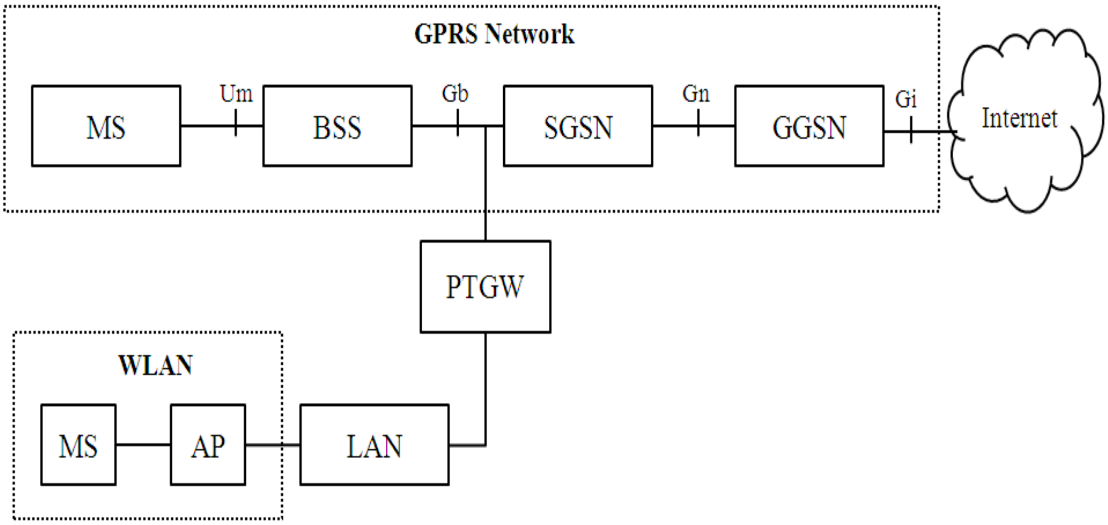

Figure 2.16: Integrated WLAN-GPRS network design [66]

networks, Protocol Translation Gateway (PTGW) has been introduced. As illustrated in Figure 2.16 [66], the WLAN network is connected with the GPRS network via PTGW at Gb interface. The main responsibility of the PTGW is to establish compatibility between networks and transparent data transfer between SGSN and ME, taking into account that different protocols are being used on either sides.

In the MS, two new functions have been introduced. Namely these are: Network Switching Function (NFC) and Ethernet Adaptation Function (EAF). The NFC basically switches the packets between the GPRS and WLAN LLC layers of MS. In addition to the ME, the EAF is also present in the PTGW. When the ME moves to the WLAN network, then the EAF encapsulates the packets so that the required information for the data routing can be sent to the PTGW via WLAN network. The EAF header contains the MAC address and Temporary Logical Link Identifier (TLLI) of the MS. Moreover, the header includes the PTGW address which is collected from the beacon messages and the AP address via it is connected with the PTGW. The peer EAF in the PTGW decapsulates the packets before sending it to the SGSN. It must be noted that, at the Gb interface the PTGW implements the BSS protocol stack. Therefore, SGSN consider it as it's another BSS. With the help of collected information from the EAF header, the PTGW records an entry, termed as Virtual MS (VMS) for each MS accessing the GPRS network via WLAN. This entry includes the TLLI, MAC address information of the MS and the AP address for the downlink and uplink LLC PDUs transportation. Despite, a sophisticated integration mechanism with the detailed signaling mechanism, no analytical or simulation analysis were presented.

Authors of [66] have analyzed and evaluated their SGSN emulator based integration protocol in [2]. It was observed that the total downward handoff latency of this integration mechanism is composed of four different latency components. Namely, these are Detection Delay (LD), VMS Delay (LV), Handoff Request Delay (LH) and Handoff Access Delay (LA). In [2], the total downward latency (Ld) has been presented by the following equation.

$$L_d = L_D + L_V + L_H + L_A \qquad (2.6)$$

On the basis of simulation results, it was observed that the total handoff latency of the PTGW integration mechanism was 540ms. In addition, in order to investigate the effect of background traffic on the handoff latency, additional users from 0 to 10 are introduced inside the WLAN and GPRS access networks. The collected statistic indicated that because of more chances of traffic collision inside the WLAN compared with the GPRS network, the handoff latency significantly increases when the background traffic is increased inside the WLAN network. Nevertheless, the upward vertical handoff analysis was not analyzed which is more crucial and highly influences the handoff latency and packet loss compared with the downward VHO case.

In [67], Tsao et al. have designed and evaluated an RNC emulator mechanism for the integration of UMTS and WLAN networks. In addition to the SGSN based tight coupling integration mechanism, MIP and gateway integration mechanisms have also been designed. In this research article, the performance of all aforementioned schemes has been analyzed in terms of handoff latency. The authors found that the Mobile IP is the easiest way to achieve the integration. Furthermore, by using the Mobile IP, networks can be deployed independently and standards are ready to use. However, the Mobile IP approach is not an appropriate solution for the real time services as the latency is too high during the handoffs. On the other hand, the handoff latency by using the gateway approach was found much lower compared to the Mobile IP approach. Although, the emulator approach is the most difficult approach among the three applied approaches and lacks flexibility because two networks are coupled tightly, on the basis of simulation results it was shown that it provides the lowest handoff latency among all of the analyzed approaches.

In order to integrate UMTS and WLAN network, the author used tight coupling integration mechanism in [68]. The authors suggested that in order to improve the network performance, in addition to the RSS, parameters like available bandwidth and network load should also be taken into consideration for handoff decision. For this, a database has been introduced inside the UMTS network which is dynamically updated with available bandwidth and network load. For the network integration, the WLAN Access Gateway (WAG) was connected with the UMTS SGSN. The simulation results demonstrated that when the wireless device performs handoff by taking the suggested parameters into the consideration unnecessary handoffs can be avoided. Moreover, the packet delivery ratio can be improved. Furthermore, similar to other tight coupling schemes, the handoff latency was found to be few hundred mili-seconds.

Another SGSN based internetworking scheme has been presented in [35]. In this tight coupling design, the Internetworking Gateway Function (GIF) connects the WLAN with the core GPRS network, as illustrated in Figure 2.17. The main responsibility of the GIF is to hide the WLAN specifics with the SGSN in a manner that the SSGN always sees the WLAN as it's another Radio Access Network (RAN). The key proposed function of this design is the WLAN Adaptation Function (WAF) which is present inside the Dual mode-MS as well as in GIF. The connection between the GIF and MS is identified with the MAC addresses, whereas, the SGSN identifies the MS with the TLLI. Therefore, in order to send the MAC and TLLI information to the GIF which is responsible to provide the interoperability between two different networks, the MS WAF encapsulates the packets with the EAF header. This header contains the MAC address information and the TLLI of the MS.

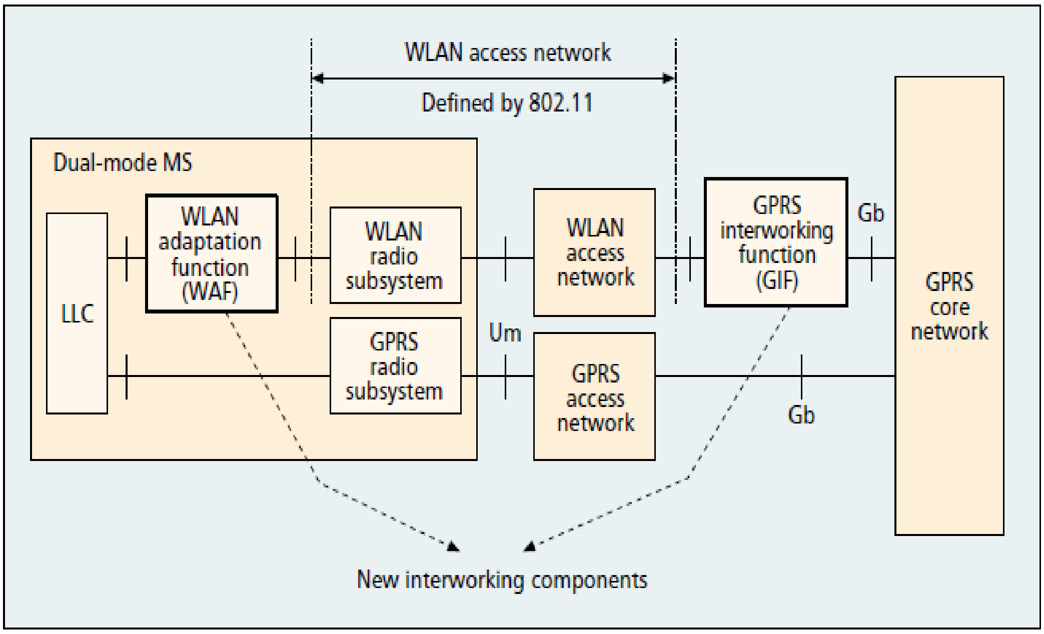

Figure 2.17: WLAN-GPRS internetworking design with the introduced network components [35]

As explained by the authors, the main advantage of this approach is the enhanced mobility management which is completely based on the existing GPRS protocols. Moreover, it protects the operators’ investment because large part of the GPRS infrastructure is used. Despite many advantages, the main disadvantage of this type of network coupling includes the enhancements in the existing SGSN component. Since, the throughput capacity of the SGSN can support thousands of low bits GPRS clients, however, may not support hundreds of high bandwidth requesters from WLAN.

In [6], the author highlighted a very interesting point which had been ignored by many research articles. It has been reported that most of the existing protocols suggested that all the ongoing session of the wireless device should be handled by one access network, either before or after the handoff. However, in case of overlay heterogeneous wireless internetworking environment, management of the separate data session with different access networks has been advocated in this article. Since, wireless clients may use several applications simultaneously; therefore, handing over the session to the most suitable network on the basis of specific QoS requirement will increase the efficiency of network resource utilization.

For the integration of UMTS and WLAN network, a SGSN based integration mechanism has been implemented. Along with the new network node, i.e., Emulated Radio Network Controller (ERNC), several additional protocol functional layers have been introduced, as illustrated in Figure 2.18 [6]. The ERNC emulates the RNC. Moreover, it monitors the AP statistics such as number of serving terminal, received signal strength, traffic load etc. The Mobile Terminal Controller (MTC) is introduced inside the MT. This functional layer is responsible to monitor the radio link and establish a priority list of preferred access networks on the basis of terminal profile. When the handoff is required, the ARNC layer requests the MTC for the measurement reports. The handoff decision is finalized on the basis of the statistics collected from the MTC and ARNC. In order to send the data traffic, when the wireless client is moved to the WLAN network, the standard GTP tunnel has been established between SGSN and ERNC. Although this mechanism provides enhanced handoff decision functionality which is based on network and terminal collected statistic, however, in

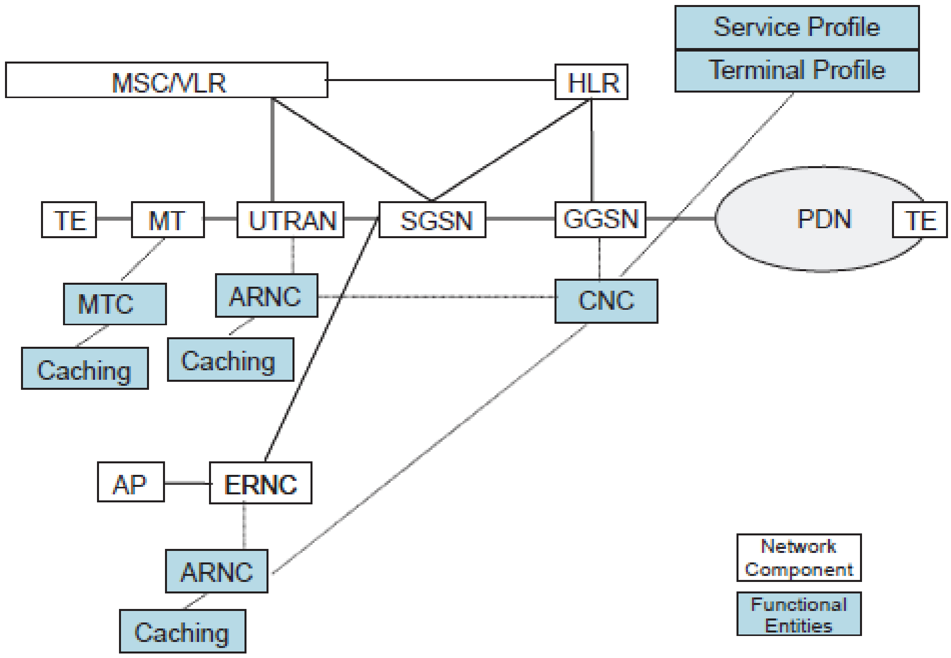

Figure 2.18: UMTS-WLAN internetworking design with the introduced network components and functional entities [6]

order to quantify the expected network resource gain of the suggested approach, no analytical or simulation results have been presented.

Authors of [6] have analyzed and evaluated their SGSN emulator based integration protocol in [69]. The proposed session based vertical handoff mechanism has been compared with two other standards: Unlicensed Mobile Access (UMA) [70] and load balancing UMA [71]. In the UMA technique, all the active sessions are switched to one of the preferred networks on user preferences only without taking into account the network parameters such as network load. In the load balancing UMA, all the active sessions are switched to one of the preferred networks; however, this mechanism also considers the current network load of the network. The network switching policy of the proposed session based VHO mechanism is similar to the load balancing UMA, however, instead of switching all the active session to one network, the switching is performed on per session based. The results demonstrated that when the network load is increased, the session based VHO mechanism provides significant improvements compared with the UMA and load balancing UMA in terms of handover and connection blocking probabilities. Although, this mechanism provides an elegant mechanism of VHO, nevertheless, the main shortcoming of this scheme is the extensive modifications in the existing networks in terms of both additional network component and additional protocol functional layers in the existing network. Moreover, no vertical handoff latency and packet loss calculations have been presented. Furthermore, signaling overheads were increased which seriously influence the network available bandwidth.

In [5], to maintain the service continuity with the seamless mobility, the authors introduced a vertical handoff protocol based on the tight coupling approach. For this, a vertical handoff management module is introduced at the Mobile Equipment (ME) that operates below the IP and above the UMTS GMM layer in the ME protocol stack,

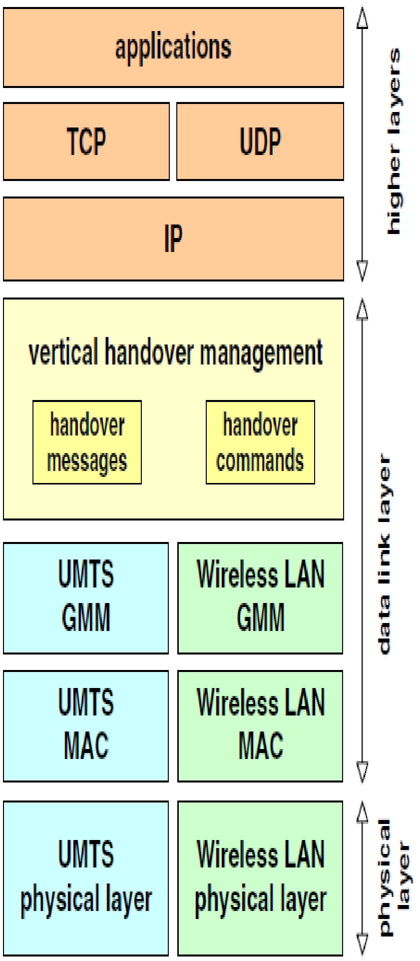

Figure 2.19: Protocol stack of the Mobile Equipment [5]

as shown in Figure 2.19. According to the 3GPP standardization, the communication between the SGSN and ME is internally addressed by the Temporary Mobile Subscriber Identity (TMSI). Moreover, a data session is established on the basis of PDP context activation. Therefore, in order to establish/continue a data session from WLAN network to external PDNs via UMTS network, the proposed scheme equips the ME with the WLAN GMM module. This WLAN GMM module works in a similar way as the UMTS GMM does. Therefore, TMSI and PDP context messages can be sent to UMTS network through WLAN network. It must be noted that the modified PDP protocol from WLAN are sent to the SGSN to avoid requesting the Radio Access Bearer (RAB) at WLAN interface.

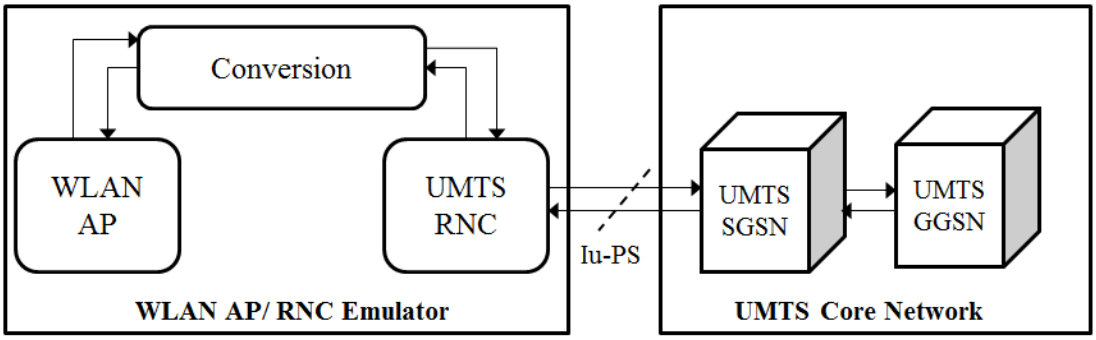

Figure 2.20: RNC emulator internetworking with the UMTS core network

In addition to the described modifications, an RNC emulator has also introduced. The design of this RNC emulator has been illustrated by the authors in [72]. Logically, this RNC emulator can functionally be divided into 3 sub-functional entities. As shown in Figure 2.20, the RNC emulator integrates the WLAN access point and standard UMTS RNC operations. Moreover, to establish the compatibility between the WLAN and UMTS networks, a conversion module has been introduced which basically performs the packet translation from WLAN format to UMTS and vice versa. The UMTS and WLAN integration is achieved by connecting the WLAN access network with the Iu-PS interface. The WLAN access point/RNC emulator is connected to the UMTS network in the similar manner as it is another UMTS Radio Access Network (RAN). A UMTS network deals with WLAN as it is its own RAN and finds no difference between UMTS and WLAN access networks.

For the data transmission from the foreign network, similar to the UMTS data management, a data tunnel has been established between SGSN and RNC emulator. With the help of detailed protocol analysis and simulation results, it was observed that the handoff blackout time (i.e., when the ME is neither connected to the UMTS nor to the WLAN network) is fast enough to support the real time services as specified by the ITU-T recommendation G.114 [73]. However, such intensive modifications in the existing networks along with the additional tunneling overheads are the main limitations of this proposed tight coupling design.

2.3.3 Application Layer Mobility Management

As discussed in the previous section, the link layer solutions are highly dependent on the underlying wireless technologies. Therefore, in order to integrate the access networks based on link layer solution extensive modifications are required in the existing network protocol stacks that increase the complexity of the internetworking solution. On the other hand, loose coupling based network layer solution i.e., MIP provides the wireless technology independent integration solution. However, it suffers with the number of problems that includes high end to end latency which severely degrades the real time applications QoS, and additional encapsulation overheads etc. In order to keep the advantages of loose coupling design and at the same time to avoid the shortcomings of MIP, research community attention was turned to find some suitable internetworking solution on higher layers.

It has widely been suggested in the literature that the loose coupling technique based on Session Initiation Protocol (SIP) can avoid many problems associated with the MIP [12, 13, 22]. SIP is an application-layer control protocol that can establish, modify and terminate multimedia sessions. The multimedia streams include audio, video, and any Internet-based mechanisms such as distributed games, shared applications, shared text editors etc. SIP defines several logical entities, namely user agents, redirect servers, proxy servers and registrars. Several wireless technical fora, such as the Wireless Internet Forum (MWIF), Third Generation Partnership Project (3GPP), and 3GPP2 are agreed on SIP utilization for the session management [22].

The authors in [74] have analyzed and compared the SIP, MIP and SCTP performance. By implementing a SIP approach, when the wireless client moves to the foreign network with an active session, in addition with new acquired IP address transmission, it sends a new session invitation to the CN. However, as described in [23, 24], the invitation message (re-INVITE message) will contain the previous call identifier as used in the original call setup. It is worth mentioning here that the re-REGISTER message is sent from the MN to the SIP server before the re-INVITE message so that the new invite message can be redirected to the new point of attachment of MN [75]. Figure 2.21 illustrates a SIP based integrated UMTS/WLAN network design. For the user agent registration, SIP registrar server is installed at UMTS core network. SIP supports both pre-call and mid-call mobility, as illustrated in Figure 2.22.

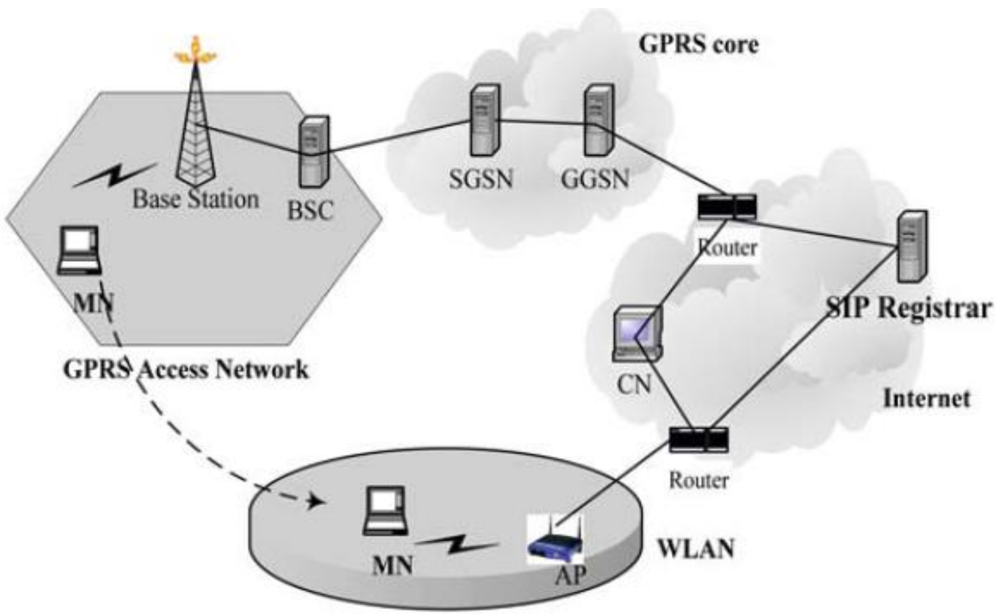

Figure 2.21: SIP based integrated UMTS and WLAN network design [74]

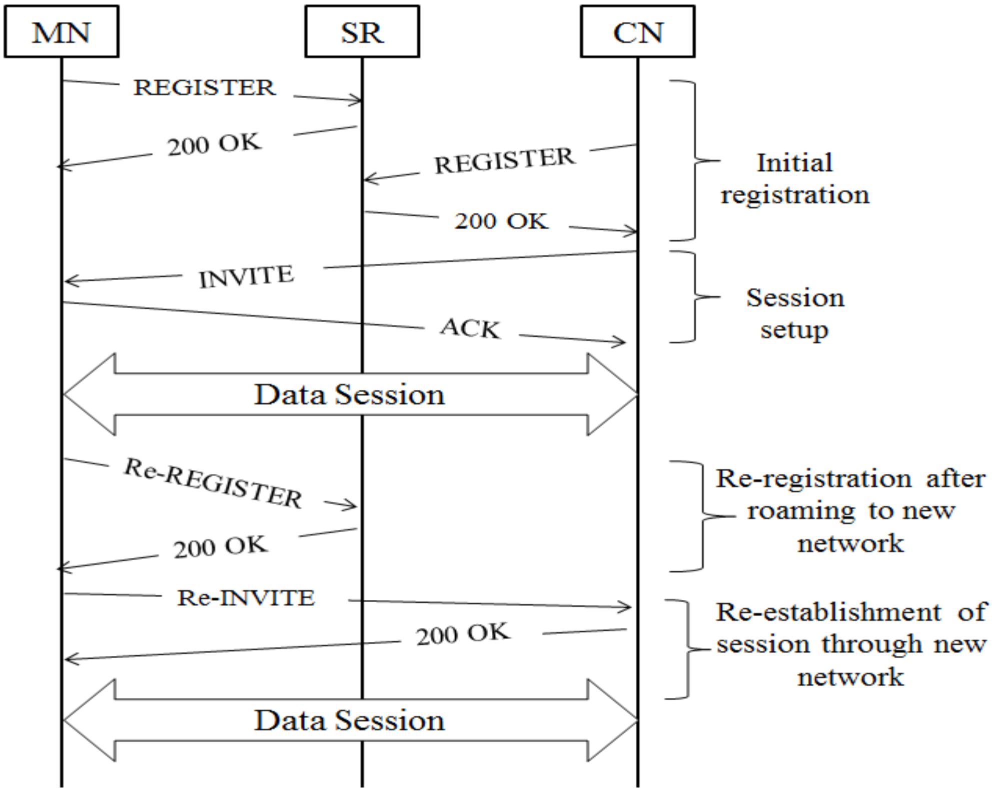

Figure 2.22: SIP pre-call and mid-call traffic signaling [74]

In case of pre-call mobility, the MN sends re-REGISTER message along with its new IP address to the SIP registrar server which is acknowledged by the 200 OK message. However, in case of mid-call mobility, the MN has also need to update the CN about its new IP address and new session parameters by sending the Re-INVITE message. In this article, two handoff scenarios have been evaluated:

1. Handoff between the Ethernet and WLAN network and

2. Handoff between GPRS and WLAN network

With the help of empirical results the authors demonstrated that, in the former case, the handoff delay in case of MIP is much higher compared to the SIP mechanism. Similarly, for the latter scenario, SIP reduces 41% handoff signaling time compared with the MIP when the user moves to the WLAN network. The reason for this reduction is straight and forward. In case of SIP, in order to perform the handoff two-way signaling delay is contributed i.e., the SIP re-INVITE message and the corresponding acknowledgement. On the other hand, the MIP requires a high number of signaling messages to perform the handoff. The high signaling traffic significantly increases the signaling latency.

In [23], Wu et al. have analyzed the SIP performance over the integrated UMTS/WLAN network. At the core UMTS network, an IMS Call Session Control Function (CSCF) has been used which provides the serving and proxy server services. In principle, the mid-call mobility management is quite similar to the MIP route optimization mechanism in which the CN start sending the data packets to the MN when received the relocation message. Both UMTS to WLAN and vice versa scenarios with the detailed signaling mechanism has been analyzed. The delays associated with the handoff process have been categorized into: wireless link and node processing delays. In [23], the handoff delay needed to switch the active connection to the UMTS and WLAN has been represented by the following expressions:

$$D_{Handoff\text{-}UMTS} = D_{Attach} + D_{PDP} + D_{SIP\text{-}UMTS} \tag{2.7}$$

$$D_{Handoff\text{-}WLAN} = D_{DHCP} + D_{SIP\text{-}WLAN} \tag{2.8}$$

Where,

D_{Attach} is the time required to perform the GMM attach with the UMTS network, D_{PDP} is the time required to activate the PDP context activation with the UMTS network and D_{DHCP} is the time required to attain the IP address from the DHCP server. In [23], the $D_{SIP\text{-}UMTS}$ and $D_{SIP\text{-}WLAN}$ delays are represented with the following expressions.

$$D_{SIP\text{-}UMTS} = D_{MH} + D_{RLP} + D_{P\text{-}CSCF} + D_{I\text{-}CSCF} + D_{S\text{-}CSCF} + \Delta_I + D_{Dest} \tag{2.9}$$

$$D_{SIP\text{-}WLAN} = D_{MH} + D_{NO\text{-}RLP} + D_{GW} + \Delta_I + D_{Dest.} \tag{2.10}$$

Where,

D_{MH}, $D_{P\text{-}CSCF}$, $D_{I\text{-}CSCF}$, $D_{S\text{-}CSCF}$, D_{Dest}, D_{GW} are the processing delay at the MH, proxy CSCF server, interrogating CSCF server ,serving CSCF server, destination node (CH), and gateway to the WLAN, respectively. The ΔI represents the Internet delay in transmitting of SIP messages. For more details on the notations, kindly refer [23].

It has been observed that the wireless component of WLAN to UMTS handoff delay (i.e., $D_{Attach} + D_{PDP} + D_{RLP}$) is about 1.5 s. However, a significant reduction in the wireless components of UMTS to WLAN handoff delay (i.e., $D_{DHCP} + D_{NO\text{-}RLP}$) has been observed. As described in this article, one of the major reasons of the high handoff latency in case of WLAN to UMTS compared with the UMTS to WLAN handoff case is more number of message transmission/reception over the error-prone and bandwidth limited UMTS wireless channel. It is worth mentioning here that the wireless link delay component deteriorates the handoff performance in several orders of magnitude compared to the node processing and transmission delays in the high speed backbone networks.

A novel approach to implement SIP by using the 3GPP IMS has been presented in [24]. For the integration of UMTS and WLAN networks, SGSN emulator has been connected with the UMTS GGSN. Similar to the tight coupling mechanism, this approach does need to acquire the new IP address when the user moves to the foreign networks. The authors highlighted some very interesting points. As shown in the Figure 2.23, the downward and upward vertical handoffs are performed in overlapped and non-overlapped coverage regions, respectively. The network selection criteria were solely based on the WLAN RSS. Therefore, in addition to the integration protocol, the non-coverage regions highly contribute to increase the session reconnection latency because of break-before-make phenomenon. The simulation results demonstrated that the handoff from UMTS to WLAN can provide the acceptable levels of service continuity. Nevertheless, as expected, the WLAN to UMTS handoff produces high vertical handoff latency. Therefore, in this case, the service continuity cannot be guaranteed.

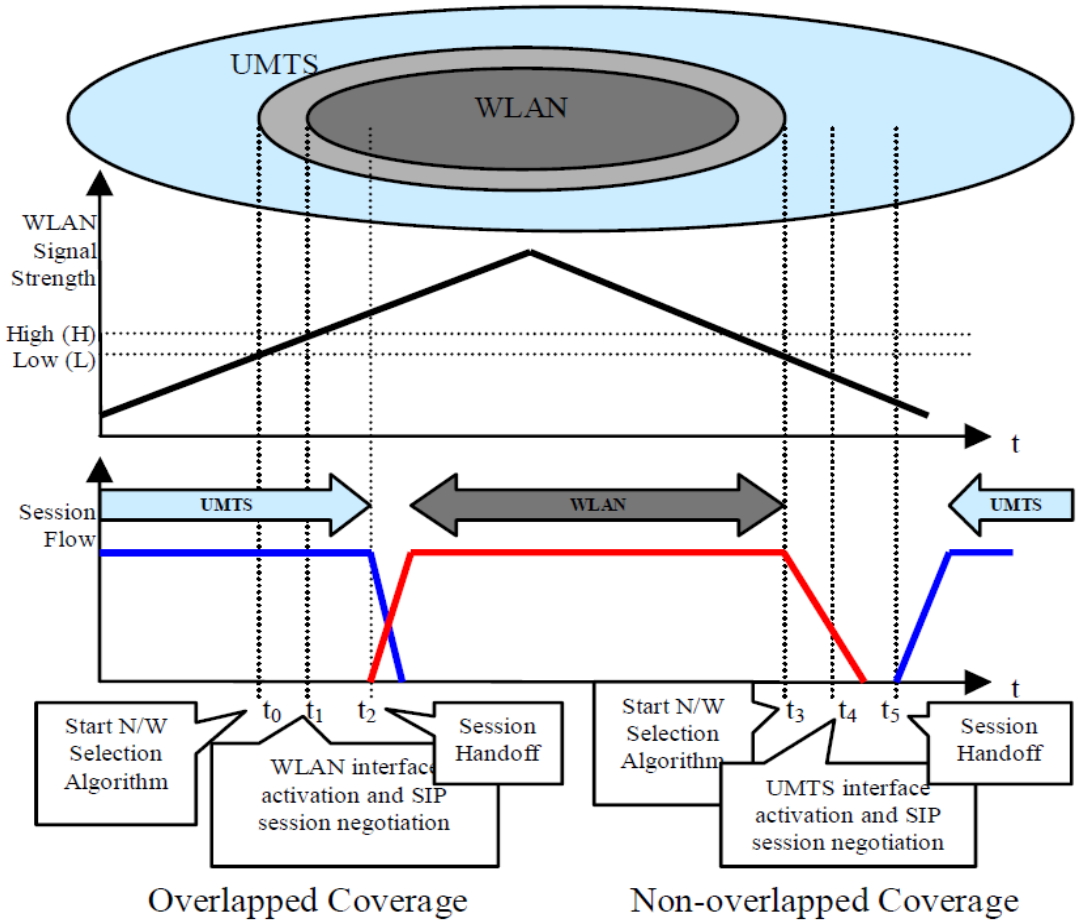

Figure 2.23: Overlapped and non-overlapped regions [24]

2.3.4 Transport Layer Mobility Management

A new transport layer protocol, Stream Control Transmission Protocol (SCTP) [76], has been accepted by the IETF as a Request for Comments (RFC). Mobility management in the transport layer is solely accomplished by use of Stream Control Transmission Protocol (SCTP) and its proposed Dynamic Address Reconfiguration (DAR) extension [77]. The SCTP with its DAR extension is called mobile SCTP (mSCTP). Actually, the DAR extension leverages the SCTP to allow the endpoints to add, delete, or change the IP address by using the ASCONF messages, while the SCTP session has already been ongoing. Such dynamic IP address modification was missing in the base SCTP version which made it less suitable for the heterogeneous internetworking environments where IP addresses are changed whenever required.

In case of MIP and SIP, home/foreign agents and SIP servers are required to be installed in the existing network, respectively. However, unlike SIP and MIP approach, in the transport layer approach only end points participate in the mobility management. Consequently, no third party interaction, additional or modified network components in the legacy network are required [25]. The salient feature of SCTP i.e., multi-homing, enables the VHO mechanism seamless. The SCTP allocates several IP addresses to the mobile user and each IP address corresponds with a particular wireless access network. Therefore, during a VHO the user can keep its session active with the pervious network by using the old IP address and at the same time establish connection with the target access network with the new assigned IP address. This eventually reduces the handoff latency in which the user is not connected to either access networks.

For the integrated UMTS/WLAN network, Ma and Yu [25] have analyzed the vertical handoff latency and throughput by using the SCTP protocol. In order to perform the handoff, three basic steps were required. Namely:

1. Add IP address (MN updates the server with the new IP address)
2. Vertical handover triggering (MN requests the server to change the primary address and receives acknowledgement)
3. Delete IP address (MN request to delete the requested IP address)

In this study, two different network configurations have been evaluated. (a) Single homing Fixed Server (FS) in which the FS contains only one IP address and (b) Dual homing FS in which the FS contains two IP addresses. For the single-homing FS case, all aforementioned basic steps to perform the handoff are executed. On the other hand for the dual-homing FS case, the MN and FS are well informed with both IP addresses of each other. Therefore, rather requesting to switch to the primary IP address to FS, the MN can directly start sending the packets to FS with the new path. Consequently, by minimizes two ASCONF messages compared with the single-homing FS case, the total handoff latency has been reduced. More precisely, the handover latency in case of single-homing configuration was found to be 533ms and 513ms for UMTS to

WLAN and WLAN to UMTS handoff cases, respectively. Nevertheless, by applying the dual homing functionality the VHO latency was reduced to 234ms and 212ms, for the movement between UMTS to WLAN and WLAN to UMTS, respectively. Besides handoff latency, the throughput parameter is found to be improved in the case of dual-homing case.

Despite, many advantages of the dual-homing approach over single-homing mechanism, the main drawback were the redundant data packets transmission from both routes during vertical handoff. Since, the redundant packets must be discarded by the receiving node; therefore, sending the duplicated packets is the wastage of scarce wireless bandwidth. Furthermore, in the current internet design, more or less every server is configured with only one IP address. Therefore, the process of up gradation of every internet server with more than one IP address would require an intensive endeavor. Therefore, as argued in [78] fixed servers with a single IP address support for mobility management is a natural choice.

In [74], the authors tested the performance of SCTP on the heterogeneous network test bed and compared its performance with SIP and Mobile IP. It has been observed that total handoff delay is least in the case of SCTP, which is followed by the SIP and highest in MIP case. Numerically, it is 31% and 55% less compared to the SIP and MIP, respectively. The main reason to execute fastest handoff by implementing the SCTP protocol was the fewer number of handoff messages requirement compared with other techniques, which consequently produces lowest handoff latency. Moreover, the transmission delay of the data packets after the handoff was evaluated. It has been found that SCTP and SIP reduce 54% and 47% packet transmission delay, respectively, compared with MIP. Since, in order to send the packets from CN to MN, the MIP redirects the packets to the home network. In home network, HA encapsulates the packets which are decapsulated by the FA in the foreign network before sending it to the MN. The encapsulation/decapsulation mechanism and long transmission path of MIP produce highest packet transmission delay.

Another effort that can effectively analysis the performance of SCTP in case of integrated UMTS/WLAN network can be found in [75]. In order to compare the SCTP performance, SIP has also been implemented on the loose coupling network design. The authors stated that the total handoff latency in case of SIP, when the MN moves from UMTS to WLAN network, is composed of:

a) WLAN authentication

b) DHCP registration

c) SIP location and

d) INVITE procedures

On the other hand, for SCTP based UMTS to WLAN handoff procedure, WLAN authentication and DHCP registration delay do not contribute to the total handoff delay because of the SCTP multi-homing feature. For the WLAN to UMTS case, SIP includes the GPRS attach and PDP context activation, however, these delay components are avoided when the SCTP is used. The simulation results show that, for the upward vertical handoff, it takes around 3-3.5 seconds by the SIP. However, because of the avoidance of GPRS attach and PDP context activation which consumes considerable amount of time, SCTP produces only few hundred mili-second delay. Similarly, for the UMTS to WLAN handoff, SCTP significantly reduces handoff delay compared to the SIP. The major reason of such improved performance is the utilization of fewer number of vertical handoff signaling messages during handoff.

In [27], in order to analyze the performance of SCTP and MIPv6, a Linux based testbed was designed. Both horizontal and vertical handoffs have been evaluated, which were referred as single-homing and dual-homing, respectively. The experimental results demonstrated that the handoff latency in case of mSCTP is approximately 2 seconds lesser than the MIPv6 for both horizontal and vertical handoff scenarios. Actually, in case of MIPv6, the MN updates the HA and CN with its current location by sending the binding updates. The MIPv6 BU mechanism significantly increases the overall handoff latency. Nevertheless, in case of mSCTP, the ADD-IP and Primary-Change operations are executed much faster that leads to attain the lower handoff latency of few hundred milli-seconds for the vertical handoff case.

2.4 Existing Protocols Evaluation and Comparison

Section 2.3 comprehensively discusses different network integration design mechanisms and suggested protocol at different TCP/IP protocol layers. It is important to notice that every proposed protocol has its own pros and corns. This section briefly demonstrates the strength and weakness of each of the presented protocol in terms of several parameters such that network implementation complexity, handoff delay, packet loss, signaling costs, modification requirements on the core network and correspondent node side etc. Table 2.2 shows the qualitative analysis of several existing mobility management protocols.

The Mobile IPv4 was the first protocol presented for the mobility management of the wireless clients moving across the different access network domains. Despite providing the advantages like session continuity, higher layer transparency, independent network deployment etc., the MIP suffers with several unavoidable drawbacks. The shortcomings include high additional encapsulation overheads, signaling cost, handoff latency and packet loss. Moreover, in order to provide the session continuity, the MIP introduces HA and FA which increase the total cost of network deployment. Furthermore, MIP does not provide a suitable solution for the data routing because of the triangular routing problem.

The MIPv6 alleviates several MIPv4 drawbacks. For example, because of the MIPv6 stateless auto-configuration feature, there is no need of FA installation at foreign networks. Therefore, only HA is required to provide the session mobility. By introducing the route optimization mechanism the MIPv6 avoids the triangular routing, since a direct data communication path between the CN and MN is established. Moreover, the MIPv6 significantly reduces the handoff latency and packet loss compared to the MIPv4, however, these parameters are still very high to realize the seamless mobility which is the prime performance parameter for NGWNs. The main reason of high handoff latency and packet loss are the high number of signaling messages and their redirection among several network components which further takes long processing time to execute specific MIPv6 task.

Table 2.2: Qualitative analysis of existing mobility management protocols

Protocols Features	MIPv4	MIPv6	Tight coupling		SIP	mSCTP
			SGSNE	RNCE		
Operational layer	Network	Network	Data link	Data link	Application	Transport
Network topology modification	Yes	Yes	Yes	Yes	Yes	No
Additional network nodes	FA/HA	HA	SGSNE	RNCE	Servers	No
Transport services	TCP /UDP	TCP /UDP	TCP /UDP	TCP /UDP	UDP	SCTP
Handoff delay	Very high	High	Low	Low	High	Low
Packet loss	Very high	High	Low	Low	High	Low
Modifications at CN or server	No	Yes	No	No	Yes	Yes
Signaling cost	High	High	Low	Low	High	low
Deployment complexity	Moderate	Moderate	High	Very high	Moderate	Low
UMTS bottle neck chances	No	No	High	High	No	No
Location Management	Yes	Yes	Yes	Yes	Yes	No

In addition, in MIPv6 route optimization case, when the mobile terminal moves to a foreign network with the ongoing session then to make the routing transparent to the upper layers and to avoid ingress filtering, the MN and IS uses home address option and type 2 routing header, respectively. The home address option with the destination extension header and type 2 routing header incurs 24 additional bytes each into the data packets for the communication [55] [79]. These additional encapsulation overheads seriously affect the network data performance and network available bandwidth. The application response time of different data traffic increases because more overhead brings more processing requirements for the intermediate nodes. Furthermore, since the bandwidth is a scarce resource, especially in wireless environment, therefore, the process of overhead append is a big obstacle to attain the high data service performance.

In order to reduce the handoff delay and packet loss, the tight coupling mechanism has widely been suggested in the literature. In the loose coupling mechanisms the access networks are connected through the internet. Therefore, for the interaction required by the integrated network nodes to execute the handoff contain long transmission delay which significantly contributes to increase the handoff latency and packet loss. By using the tight coupling schemes the transmission cost is reduced because the integrating network is directly to the UMTS core network. In addition to the transmission cost, the fewer number of signaling messages are required to execute the handoff. As a result, the handoff delay and packet loss are reduced, thus, improved seamless mobility features are attained. Furthermore, UMTS core network resources, subscriber database, billing system and authentication mechanism can be reused. This reuse of resources leads to low cost network deployment.

However, the main limitation of the tight coupling mechanism is the high operational complexities compared with the loose coupling technique. This happens because the internetworking protocols are highly inspired by the existing UMTS mobility management protocols. Therefore, in order to establish the compatibility among different networks intensive modifications at the integrating network interface is required so that the integrating network can implement all the UMTS protocols required at the UMTS access or core network. Moreover, for protocols’ conversion, UMTS protocol’s implementation on the integrating network side and data routing between UMTS and WLAN networks, additional network emulators (SGSN and RNC emulator)/gateway devices are required. Since, the internet service providers/ telecommunication operators all over the world have already invested an enormous amount to the fully functional legacy networks; significant modifications in existing protocols and the introduction of additional mobility management components would not easily be accepted. Because of high dependence on the access technology, if more and more networks are integrated together, more gateways/emulators would be required for the protocol conversions. Furthermore, as the data traffic of WLAN traverse via the UMTS network, they potentially create a bottleneck in the UMTS network. Therefore, in order to avoid this, modifications in the existing UMTS nodes are required to sustain high traffic loads.

In UMTS network, for the data transportation, it is required to create a GTP tunnel from the GGSN to SGSN and SGSN to RNC. Similarly, as illustrated by the literature same convention has been followed by the researchers for the integrated UMTS/WLAN network and a tunnel between the GGSN to SGSNE or RNC emulator was established. The GTP tunnel incurs additional 12 bytes overhead into each data packet for transportation [80-83]. Such overheads increase the network nodes processing and, hence, increase the application response time.

Despite many strong features such as no modification requirements at the CN and independent network deployment and traffic engineering, some drawbacks of the SIP protocol make it less attractive for the internet service providers to adopt it as a complete mobility management solution for the NGNs. For example, when a user moves across different networks, every time when it connects to a new network, it has to acquire an IP address to continue its session. Since, the SIP is operated over application layer; therefore, SIP cannot re-establish a broken TCP connection because of changes in the network layer address during/after handoff [7, 12, 22]. Therefore, as reported by Politis et al. [22], SIP based mobility management applies only to the real-time communications over User Datagram Protocol (UDP). In consequence, even addressing the personal and service mobility efficiently, SIP does not provide a transparent terminal mobility [22]. In addition, SIP based mobility management can only be applied to the SIP aware applications [37]. This simply suggests that if SIP is applied, a comprehensive drafting in the existing applications has to be undertaken in order to be mobility-aware [7]. Since SIP uses text based messages, therefore, messages sizes are much more generous. As a result, compared to MIP approach, SIP produces much higher signaling load on the network [84].

In addition, similar to the MIP and tight coupling mechanisms, SIP introduces additional mobility management components (SIP servers) in the existing networks. Although, SIP requires fewer numbers of signaling messages which execute faster handoff than MIPv6, however, still the vertical handoff latency is quite high to provide the uninterrupted real-time service experience to the end user.

Despite many strong features compared with the MIP, tight coupling and SIP such as no modification in the existing network topology, low handoff delay and packet loss etc., the main obstacle in the deployment of the mSCTP is the dependency over the SCTP transport services. The SCTP is designed to eventually replace the TCP and UDP transport layer services. Even though, the SCTP is many aspects better than TCP and UDP, however, implementation of SCTP based transportation services are quite less. This is because of the fact that currently almost every device software implementation is based on TCP and UDP. The SCTP selection for the transport service will obviously require reconfiguration of existing devices. The devices over which TCP and UDP protocols are being operated as the transport layer protocols are literally beyond to any counting mechanism. A global rollback of the widely deployed TCP and UDP based device is extremely difficult, if not impossible [25].

Furthermore, another very important mobility management parameter that is missed when mSCTP solely deal with the mobility management is “location management” [26, 27]. Since, the SCTP is an end-to-end protocol in which only the end points participates and updated with the MN relocation. Consequently, if the MN moves to the foreign network then new session establishment request cannot be facilitated by the home network, since it would be unaware with the MN relocation. Therefore, in order to provide the location management, mSCTP can be used along with the MIP or SIP [27].

2.5 Chapter Summary

This chapter discussed the evaluation of wireless networks from 1G to 4G whilst highlighting the need to switch from one wireless generation to another. Since, the limitations of the existing network were indicating the need of unified integrated wireless heterogeneous access networks, therefore, the 4G network concepts, features, and its requirements were discussed in detail. Moreover, the review of the background work also summarized the UMTS and WLAN networks, their topologies, functional entities and protocols which were important to develop the understanding of the basic concept and functions of integrated UMTS/WLAN network.

Section 2.3 comprehensively discussed several contemporary mobility management protocols for the heterogeneous wireless networks. Section 2.4 highlighted that existing techniques are still lacking to meet the requirements of 4G networks. The most important unsolved issues were high handoff latency and packet loss, intensive modification requirements in the existing networks in terms of both i.e., either introduction of additional network nodes or alteration of existing protocol stacks, and high additional protocol overheads when the wireless client moves to the foreign network which eventually deteriorates the data session by increasing the application response time.

Therefore, in order to address the shortcomings of the contemporary internetworking approach, a Seamless Vertical Handoff Protocol (SVHOP) is proposed. The next chapter presents the proposed SVHOP's design considerations, implementations and evaluation, and detailed upward and downward vertical handoff performance analysis by comparing it with the benchmark protocols.

CHAPTER 3

RESEARCH METHODOLOGY

3.1 Chapter Overview

In the previous chapter, the contributions made by different researchers and standardization bodies for the integration of UMTS/WLAN network have been discussed in detail. According to the analysis and collected statistics from literature, it is quite manifested that none of the existing protocol can be considered as the complete viable solution for the network integration. This simply indicates that here there are a lot more efforts required to realize the NGN networks practically.

In order to bridge the gap between the objective of the 4G networks and available solutions for the integrated UMTS/WLAN network, in this chapter a Seamless Vertical Handoff Protocol (SVHOP) is proposed. The design considerations, network topological design ranging from the core network to the wireless client, implementation and evaluation of the proposed SVHOP is presented. The SVHOP addresses many issues which are crucial in defining the integration of the UMTS/WLAN network and those were not tackled by the many existing mobility protocols.

3.2 Research Process Flow

Figure 3.1 depicts the flow of the processes in this research.

Phase 1: A comprehensive literature review in the area of mobility management in NGWN was carried out. Various domains such as integrated network design that

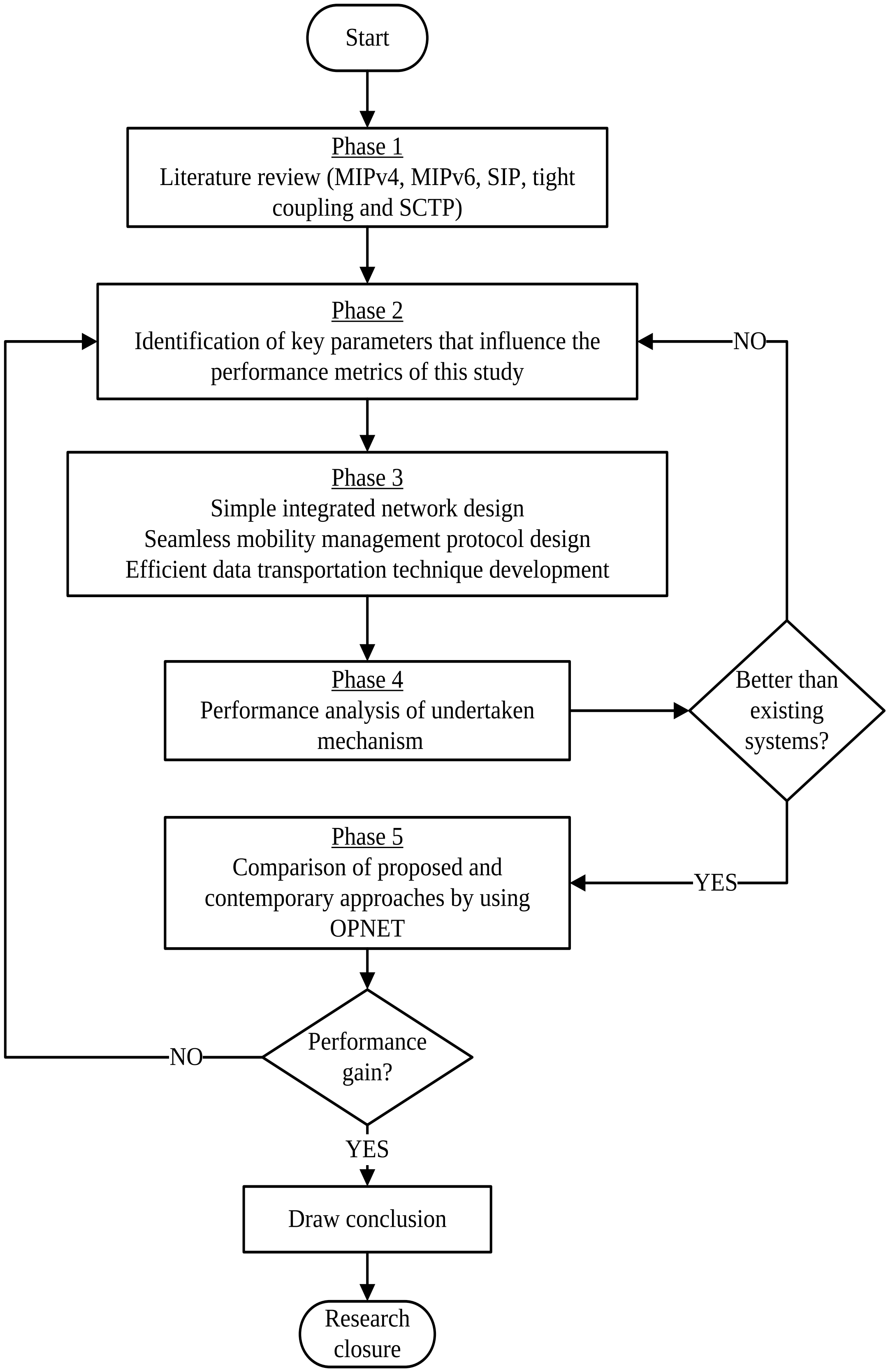

Figure 3.1: Research process flow

includes tight coupling and loose coupling internetworking architecture design and internetworking protocols such as MIPv4, MIPv6, SIP, SCTP etc., were analyzed.

Phase 2: After a thorough analysis of literature review in phase 1, during this phase:

- The gap between the state of the art solutions and desired objectives of wireless heterogeneous networks are identified.

- The key issues that fall under the scope of this research work were isolated. Moreover, parameters that influence the performance metrics were studied.

Phase 3: Having identified the shortcomings of the existing integration network design and internetworking protocols, in order to provide the seamless mobility management along with the minimum modification to the existing network design, following actions were performed:

- Devise a seamless vertical handoff protocol that ensures very low handoff latency and packet loss while the wireless client is moving across the UMTS and WLAN access networks.

- Design an integrated UMTS/WLAN network design that ensures the minimum modification to the existing UMTS and WLAN networks.

- Develop an additional protocol overhead free data transportations mechanism so that the bandwidth intensive applications such as FTP, HTTP etc., can be downloaded and uploaded faster.

Phase 4: During this phase, detailed performance analysis of the proposed mechanism with the contemporary approaches is evaluated. At every step it was closely monitored whether the proposed mechanism is simpler and efficient compared to the existing mechanisms. This step suggested either move to the next phase or return to the phase 2 if further enhancements are required.

Phase 5: The proposed integration network design and internetworking protocol is compared with the most prominent contemporary techniques such as MIPv6 and tight coupling mechanism by using the OPNET Modeler tool. The performance gain leads to draw the conclusion, whereas, performance setback suggested to move back to phase 2.

In order to share the new research prospects and horizons, framework, designed methodology, performance gain compared to the contemporary internetworking techniques, and to collect suggestion from research community working in the field of NGWN, several research articles were published during the course of this research work. A list of publications corresponding to this research work is listed at the end of this dissertation.

3.3 Description of the Proposed SVHOP

The Proposed mechanism, called Seamless VHO Protocol (SVHOP) mainly accomplishes three fundamental requirements of the NGWHNs. Namely:

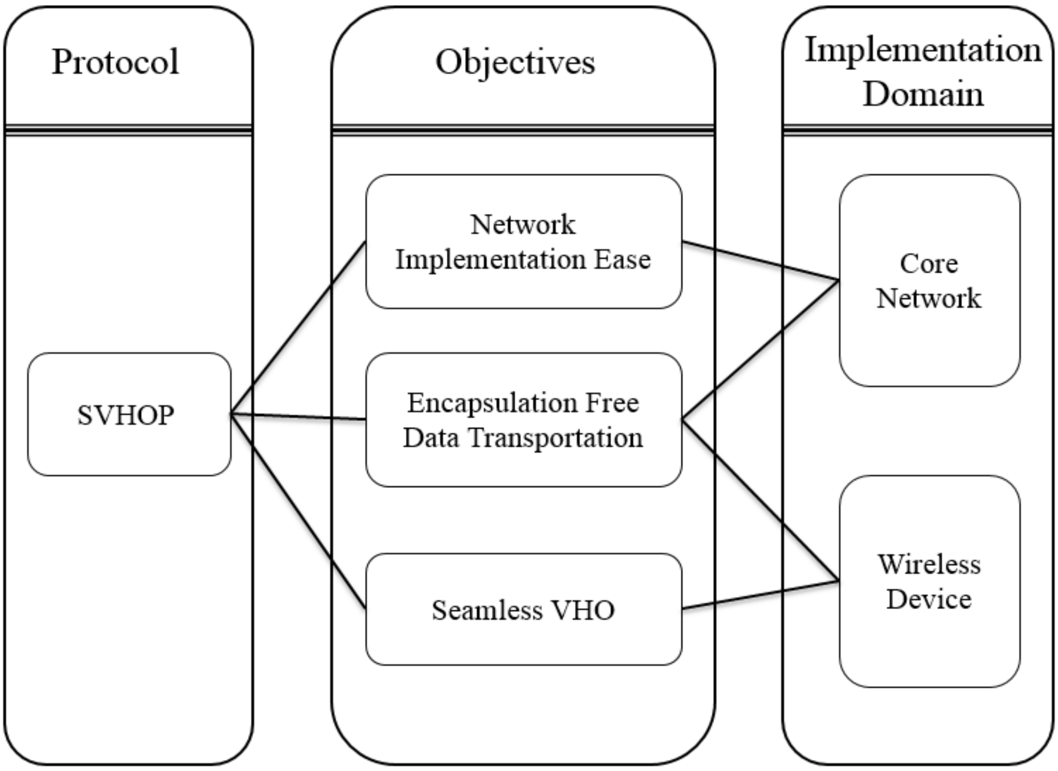

Figure 3.2: A basic sketch of SVHOP objectives and implementation domains

- Seamlessness of VHOs
- Ease of network implementation
- Additional protocol overhead free data transportation

Figure 3.2, illustrates the basic sketch of SVHOP objectives and implementation domains. The implementation domains to achieve the targeted objectives ranging from the core network to the wireless client device. The following sub-sections explain each objective with the corresponding amendments into the integrated UMTS/WLAN network design.

3.3.1 Seamless Mobility

The seamless mobility is considered as the most important performance metric while the wireless client is roaming across the integrated wireless network access network. A handover is considered seamless when it provides both smooth (no or very little packet loss) and fast (low latency) switching of active connection between the heterogeneous access networks. One of the main objectives of the Next Generation Wireless Heterogeneous Networks (NGWHN) is to ensure the seamless mobility. However, the literature illustrates that most of the existing internetworking techniques are lacking in providing the seamless mobility feature. In existing mobility management protocols, the vertical handoff latency was found to be within the ranges of hundreds of milliseconds to few thousands of milliseconds. According to the International Telecommunication Union (ITU) in their recommendation G.114, it has been suggested that that the one way end-to-end delay for the real time services should not exceed 150ms.

In order to maintain the seamless mobility during vertical handoff, a Dual Mode Mobile Terminal (DMMT) is introduced in this research study. The DMMT is capable to perform the network layer proactive vertical handoff by predicting the upcoming vertical handoff events on the basis of data link layer RSS hints. In addition, to decrease the VHO latency and packet loss, the SVHOP implements less signaling and node processing cost mechanism by adopting the multi-homing features in the wireless device. Because of parallel data transmission during the handoff, the efficient VHO is performed by sending most of the VHOs signaling messages to the target network in parallel with the already established data session with the previous network. Thus, only few signaling messages influence the overall VHO performance. Section 3.3.1.1 discusses the signaling and the node processing cost impacts on the handoff delay and packet loss. Section 3.5.1.1 explains the hypothesis by which the concept of multi-homing is being adopted into the proposed design. The proposed vertical handoff mechanism, by implementing the aforementioned metrics, is presented in Section 3.7.

3.3.1.1 Signaling and Node Processing Cost

The signaling and node process cost are the metrics that highly influence the vertical handoff latency and packet loss. Analytically, the total vertical handoff delay ($D_{VHO,\,x}$) of any "x" protocol can be expressed by summing the total signaling (*Scost*) and nodes processing (*Pcost*) cost.

$$D_{VHO,\,x} = S_{cost} + P_{cost} \tag{3.1}$$

The *Scost* is proportional to the distance between communicating nodes and type of media signals traverse to. The *Pcost* depends on the process load and functionalities of the network component. The above equation suggests that if more signals are exchanged and more nodes are participated to perform a handoff then the HO delay will be higher. Similarly, if less signaling is performed and less number of nodes taking part in the handoff process, a faster handoff can be executed. Mathematically, the packet loss during VHO period can be expressed as:

$$Packet\ loss = D_{VHO,\,x} * \lambda_p \tag{3.2}$$

Here, the λp represents the packet arrival rate. It is quite obvious from the above equation that the packet loss during vertical handoff are directly depends upon the vertical handoff latency. This is because, during the handoff period when the wireless client is not connected to any network, all the packets send to the wireless client will be dropped. Having identified the significance of low signaling and node process cost on handoff latency and packet loss, the proposed vertical handoff mechanism avoids high interaction and processing of signaling traffic by the network nodes during handoff period. Section 3.7 discuss in detail that how the SVHOP efficiently address the vertical handoff events in an integrated UMTS/WLAN network.

3.3.2 Ease of Network Implementation

Along with seamless VHOs, the network implementation ease to support seamless mobility is one of the most important parameters to bring the integrated network architecture design into reality. The literature dictates that the tight coupling mechanism requires major modification in the existing network protocols. Moreover, tight coupling introduces additional network component emulators to establish compatibility among different networks. Therefore, despite providing the low handoff latency and packet loss during handoff, the tight coupling is not been preferred because it is one of the most complex approaches to be implemented for an integrated network design. On the other hand, loose coupling techniques such as MIP and SIP required Foreign Agent/Home Agent (FA/HA) and SIP servers to be installed in the core network to perform the vertical handoffs, respectively. Since, the internet service providers/telecommunication operators all over the world have already invested heavily to the fully functional legacy networks; significant modifications to existing protocols, additional mobility management components or a global roll out with a new technology would not be an easy task. Therefore, not only the seamless mobility but the network implementation ease must be considered for any practical solution for the NGN networks.

To attain the network implementation ease, the proposed network framework requires only few alterations in the existing UMTS and WLAN network design. More precisely, only the UMTS GGSN component is needed to be upgraded with the suggested Convergence Layer (CL) module. A detailed discussion on the integrated UMTS/WLAN network topological design and CL operations are presented in Section 3.4.

3.3.3 Additional Protocol Overhead Free Data Transportation

The bandwidth is the scarce resource in a wireless environment. Therefore, any suitable integration mechanism must ensure that the data transportation mechanism has as low overheads as possible. Therefore, in order to provide the optimal data performance, it is desired that any sophisticated integration protocol must avoid inserting its own additional overhead information in the data traffic. Most of the existing techniques including MIPv6 and tight coupling add APOs to the Packet Data Units (PDUs). The process of overhead appending is a big obstacle to attain high data service performance. The PDU encapsulation is required when the wireless device is moved to the foreign network from the home network. Because of the proposed integrated network design and IP swapping mechanism, additional protocol overhead free data transportation is achieved. Section 3.4 represents the proposed network design and Section 3.4.2 demonstrates the proposed IP swapping mechanism to attain the APO free data transportation in the foreign network.

3.4 Proposed Integrated UMTS/WLAN Network Topological Architecture

In the proposed UMTS/WLAN network design, the network implementation ease is achieved by introducing an integration mechanism that does not require protocol alteration or introduction of additional network components in existing UMTS and WLAN networks. The proposed network design operates when the WLAN AP is connected with the UMTS GGSN. However, unlike the techniques based on tight coupling mechanism discussed in chapter 2, which connect WLAN AP to the GGSN through the Gn interface and operations are performed on the data link layer, the SVHOP operates when WLAN AP is connected with the GGSN at Gi interface, as shown in the Figure 3.3. When the networks are connected by using the Gi interface, there is no need to establish the link layer compatibility between the integrating networks. This is because by such internetworking mechanism the integrating networks are operated independently with each other and network layer operations are implemented for the communication between the networks. The network layer operations decouple the access technologies. Consequently, provides independence of underlying access technologies. Hence, offers independent network deployment and protocol engineering of different integrating networks.

In the proposed network design, the integrated UMTS/WLAN network is setup in a manner that both the integrated access networks are in different IP domains. As a result, when the UMTS wireless device moves to the WLAN network, it requests for the new WLAN IP address. Since, both networks are divided on the basis of IP addresses, therefore, instead of the WLAN access network appears as another UMTS RAN like in the tight coupling integration mechanism, the WLAN network act as an external PDN to the UMTS network.

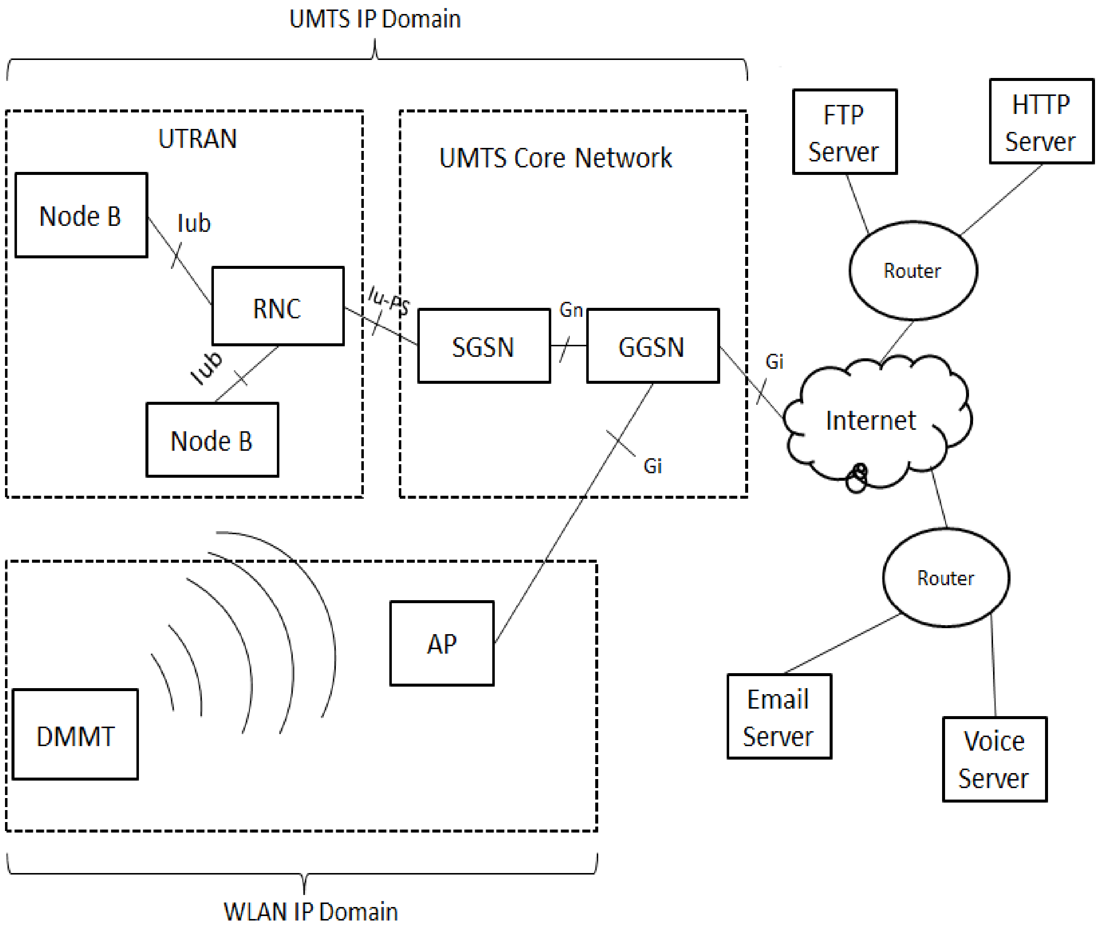

Figure 3.3: Proposed integrated UMTS/WLAN network topology

In the tight coupling mechanism the WLAN data passes by the UMTS core network and a lot of efforts are needed to:

(a) Modify the UMTS network architecture

(b) Alter the network interfaces and

(c) Install the network emulators to establish compatibility with the integrating WLAN network.

On the other hand, several loose coupling techniques such as MIP and SIP requires Foreign Agent/Home Agent (FA/HA) and SIP servers to be installed in the core network for mobility management, respectively. It is worth recalling here that the internet service providers/ telecommunication operators all over the world have already invested an enormous capital to the fully functional legacy networks. Therefore, extensive alteration in the network interfaces by implementing significant modifications to existing protocols, additional mobility management components etc., would not be appreciated. Therefore, it is believed that the network design must reuse the existing network components functionalities as much as possible for the mobility management as well as for the data transportation in the proposed integrated UMTS/WLAN network.

As illustrated in the Figure 3.3, the proposed network design does not introduce any network emulator or mobility management entity in the existing UMTS or WLAN network. Moreover, no interface alteration is required to be performed in the proposed network design because of network layer operations. For the efficient mobility management and data transportation, most of the UMTS and WLAN network components are reused without any modification. Nevertheless, some minor adjustments in the UMTS GGSN and mobile terminal are needed by introducing the suggested Convergence Layer (CL). The CL remains transparent when the session is ongoing from UMTS network. Nevertheless, it performs IP swapping mechanism when the wireless client moves with the ongoing data session from UMTS to WLAN network. The DMMT CL layer and its functionalities are discussed in detail in Section 3.5.4.

In the proposed integrated UMTS/WLAN network design, when the UMTS and WLAN clients are located in the UMTS cell and WLAN AP coverage regions, respectively, and accessing the Internet Servers (ISs) then UMTS and WLAN network will be operated completely independent and transparent to each other. No protocol or traffic engineering alteration would be required. In this case, both networks will operate as if no integration is performed. The UMTS user will send requests and receive responses to ISs through its UMTS core network without any modified procedure. Similarly, the WLAN client will send requests and receive responses via its WLAN network. At this instance, GGSN appears as a normal router which is routing the packets to/from the internet servers to the WLAN network.

However, in case of a roaming user minor modifications in the existing GGSN mechanism are required, which has virtually no significance when compared with the overall network performance improvements and the reduction in the telecommunication service providers' capital investment.

The proposed network topology is called provides the network implementation ease by avoiding:

- Significant modifications in the existing network architectures and protocol stacks and

- Introduction of additional network components in the integrated UMTS/WLAN network

The simplicity, straightforwardness and ease of implementation of the proposed network design permit the network service providers to take advantage of other providers' existing deployed networks. As no emulator/gateway/mobility agents/servers are required for the integration, consequently, integration does not require major capital investments.

3.4.1 Data Transportation in the UMTS Network

In the proposed integrated UMTS/WLAN network architecture, when the wireless client sends the PDUs from the UTRAN, the GGSN operates the session by using the traditional UMTS traffic engineering mechanism. In order to understand the data transportation mechanism between the wireless client and external packet networks and the UMTS nodes operation on the data packets, let us assume that the wireless client send/receive the data packets to/from the internet servers. Following the UMTS specification [85], the GPRS Tunneling Protocol (GTP) tunnel is required to be established for the data transportation from UTRAN to the UMTS core network or vice versa.

The GTP is used for both signaling and data transfer. The signaling GTP is known as GTP-c, whereas, the GTP-u is used for the data transportation. The GTP-c is a tunnel control and management protocol. It is used to create, modify, and delete the tunnel by using the PDP context request, PDP context update, and PDP context delete procedures, respectively. The GTP-u is implemented at the Iu and Gn interfaces that exist between the UTRAN and the SGSN, and between the SGSN and GGSN, respectively [85]. In the proposed network design only the PS domain is considered, therefore, when the UE sends the data packets to the IS, the GTP tunnel is created between RNC and SGSN, and between SGSN and GGSN, as illustrated in Figure 3.4.

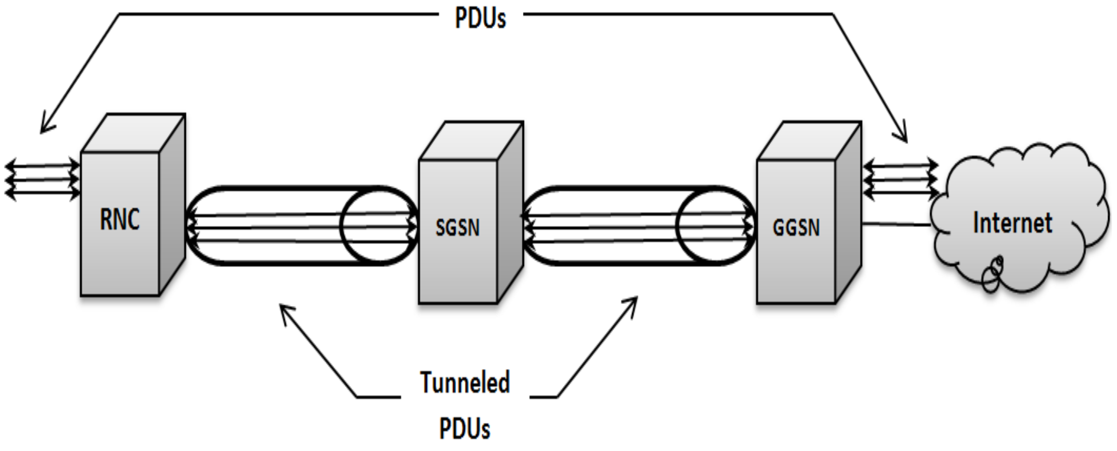

Figure 3.4: GTP tunnel establishment for the data transportation inside the UMTS network

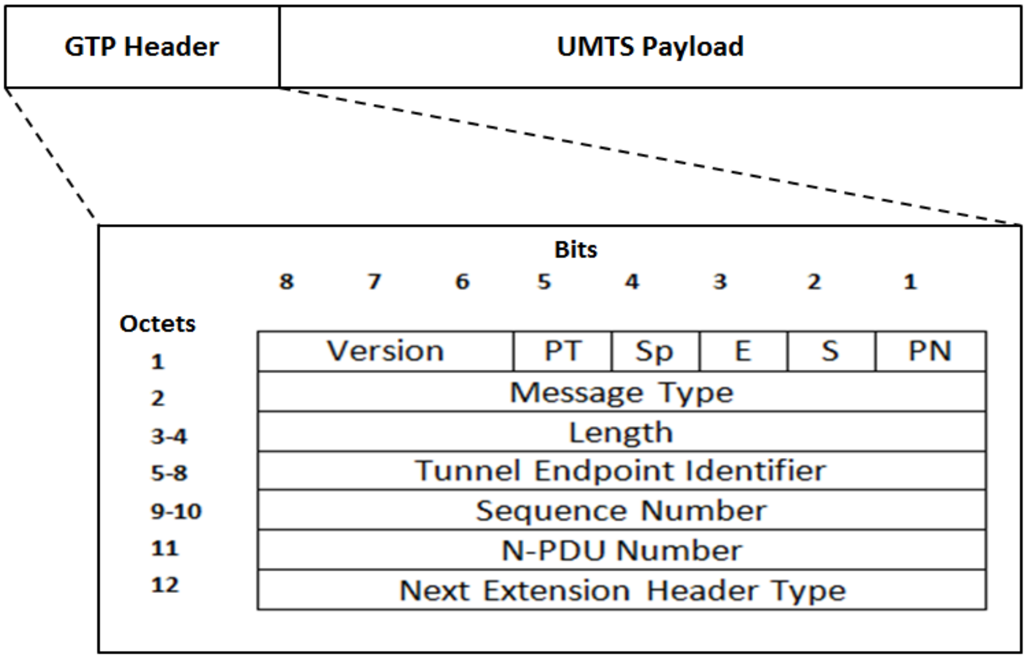

Figure 3.5: GTP header format

The GTP-u attaches a GTP header with the PDUs for the data transportation inside the UMTS network, as illustrated in Figure 3.5. Besides tunneling, the GTP-u also ensures the in-sequence delivery of the data packets. The details of the GTP header field can be found in Appendix C.

When the tunneled data is received by the GGSN from the SGSN, the GGSN IP module sends the encapsulated data packets to the GGSN GTP module. The GTP module decapsulates the data packets and send it back to the IP module as illustrated in the Figure 3.6. Since, the destination is the internet server, therefore, the IP module route the packets to the internet cloud via CL. In the reverse direction, when the IS sends the data packets to the UE, the IP module receives the data packets from the IS via CL. The IP module checks the destination address of the data packets. Since, the UMTS UE is the destination, therefore, the IP module sends the data packets to the GTP module for GTP encapsulation. The encapsulated data is sent back to the IP module, which is forwarded to the SGSN. It must be noted that the CL module remains transparent to all the operations until the UE sending and receiving the data from UMTS cell. Nevertheless, when the UE moves to the foreign network, the CL performs IP swapping mechanism to avoid additional overheads which is crucial to save the scarce wireless bandwidth.

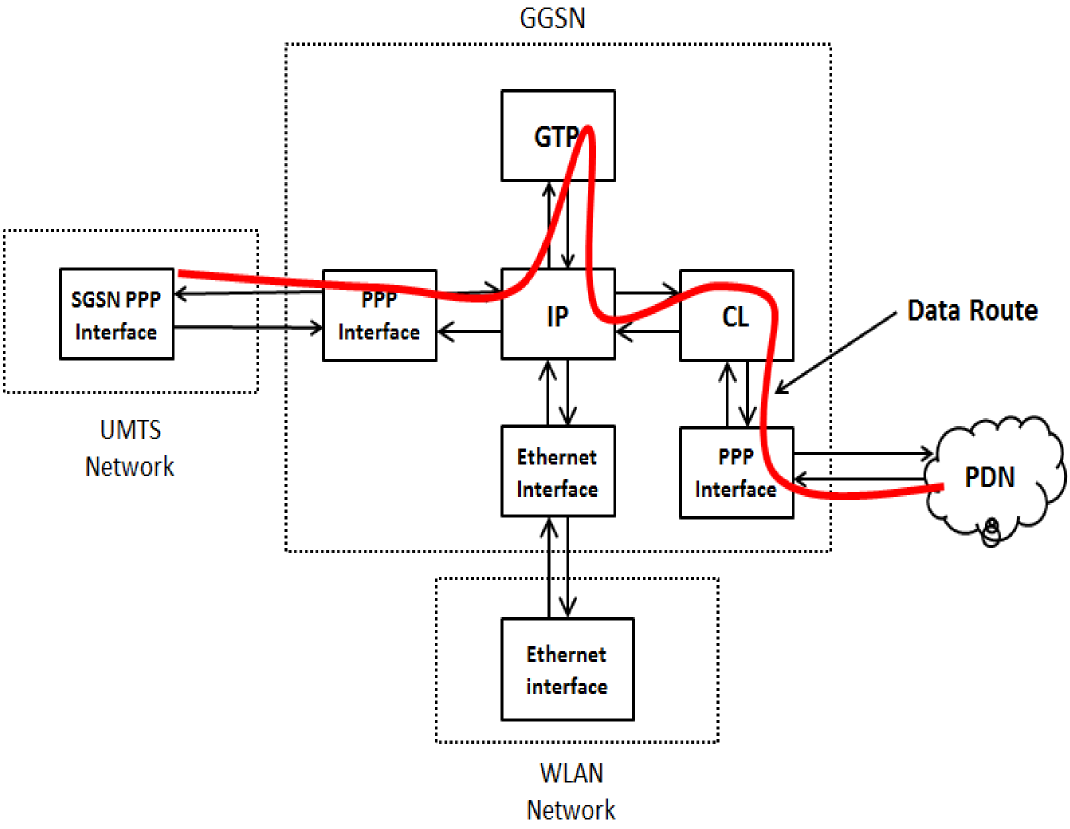

Figure 3.6: Data routing and processing inside the GGSN for UMTS data traffic

3.4.2 APO Free Data Transportation Mechanism in the Foreign Network

To understand the impacts and significance of the proposed APO free PDUs transmission/reception in an integrated heterogeneous wireless internetworking environment, let us consider a wireless client is moving from UMTS to WLAN network.

In MIPv6 route optimization case, when the mobile terminal moves to a foreign network with the ongoing session or initiates a data session from a foreign network then to make the routing transparent to the upper layers and to avoid ingress filtering, the MN and IS uses home address option and type 2 routing header, respectively. The home address option with the destination extension header and type

2 routing header incurs 24 additional bytes each into the data packets for the communication [55, 79]. Although the 802.11b network provides a data rate of 11Mbps, the inherent MIPv6 overhead severely influenced the network available bandwidth/maximum throughput. Bandwidth is a scarce resource, especially in a wireless environment. Therefore, the process of overhead appending is a big obstacle to attain high data service performance.

In the tight coupling mechanism the Gn and the Iu-PS interface is the network integration point for SGSN emulator and RNC emulator cases, respectively. Consequently, the WLAN network appears as another UMTS RAN to the UMTS core network. Hence, the UMTS network protocols for the data transportation is implemented in which the GTP tunnel is required to be established from/to RNC to/from SGSN, and from/to SGSN to/from GGSN. In case of RNC emulator tight coupling mechanism, the data from the WLAN access network is encapsulated by the RNC emulator by implementing the GTP tunneling mechanism. The tunneled data passes through the SGSN and finally decapsulated by the GGSN before sending to the external packet networks. In the reverse case when the data are coming from the external network, the GGSN encapsulate the data packets which are forwarded to the RNC emulator via SGSN. The RNC emulator decapsulates the packets before forwarding them to the wireless clients. Consequently, similar to the MIPv6 mechanism, higher number of additional overheads would be required to send the PDUs to/from a wireless device. However, the performance degradation is comparatively lesser in case of tight coupling because the tunneled data is never transmitted to the air interface.

In contrast, the proposed mechanism avoids any amendment of additional overhead information in the PDUs or tunneling establishment between the network nodes. The tunneling overheads are avoided because of the proposed topological design in which the WLAN network is considered as an external PDN to the UMTS network. Therefore, when the wireless client sends the PDUs, then it passes through the AP to the GGSN in the same way as two routers send/receive packets from each other. Moreover, no interaction of GTP module would be required in order to send/receive the data to/from internet servers.

In the proposed mechanism, when the DMMT roams from UMTS to WLAN network, to avoid the ingress filtering it has to acquire and use the new IP address from the current access network to continue its session. Now, the session can be sustained by the current IP address, however, the upper layer and the correspondent node recognizes the DMMT session with the pervious IP address i.e., UMTS_IP. The CL module of the DMMT and the GGSN avoid the upper layer and correspondent node transparency issues, respectively. The CL operations in the DMMT have been discussed in Section 3.5.4.

Message type	Src_IP addr.	Dest_IP addr.	UMTS_IP	WLAN_IP

Figure 3.7: WLAN Association Notification message format

To avoid the IP address conflict by maintaining the correspondent node's transparency, the DMMT sends the *WLAN Association Notification* message to the GGSN. Actually, this message contains the IP address assigned by the UMTS network and the IP address acquired from the WLAN network, as shown in Figure 3.7. On receipt of the *WLAN Association Notification* message, the GGSN updates its Routing Notification Table (RNT) at the CL. The RNT contains the UMTS IP address with its corresponding WLAN IP address entry, as illustrated in Figure 3.8. By this double entry IP address message the GGSN will be notified that this is the same terminal, who initiated the session in the UTRAN with the dynamically assigned UMTS IP address, now moving to the WLAN with the notified WLAN IP address.

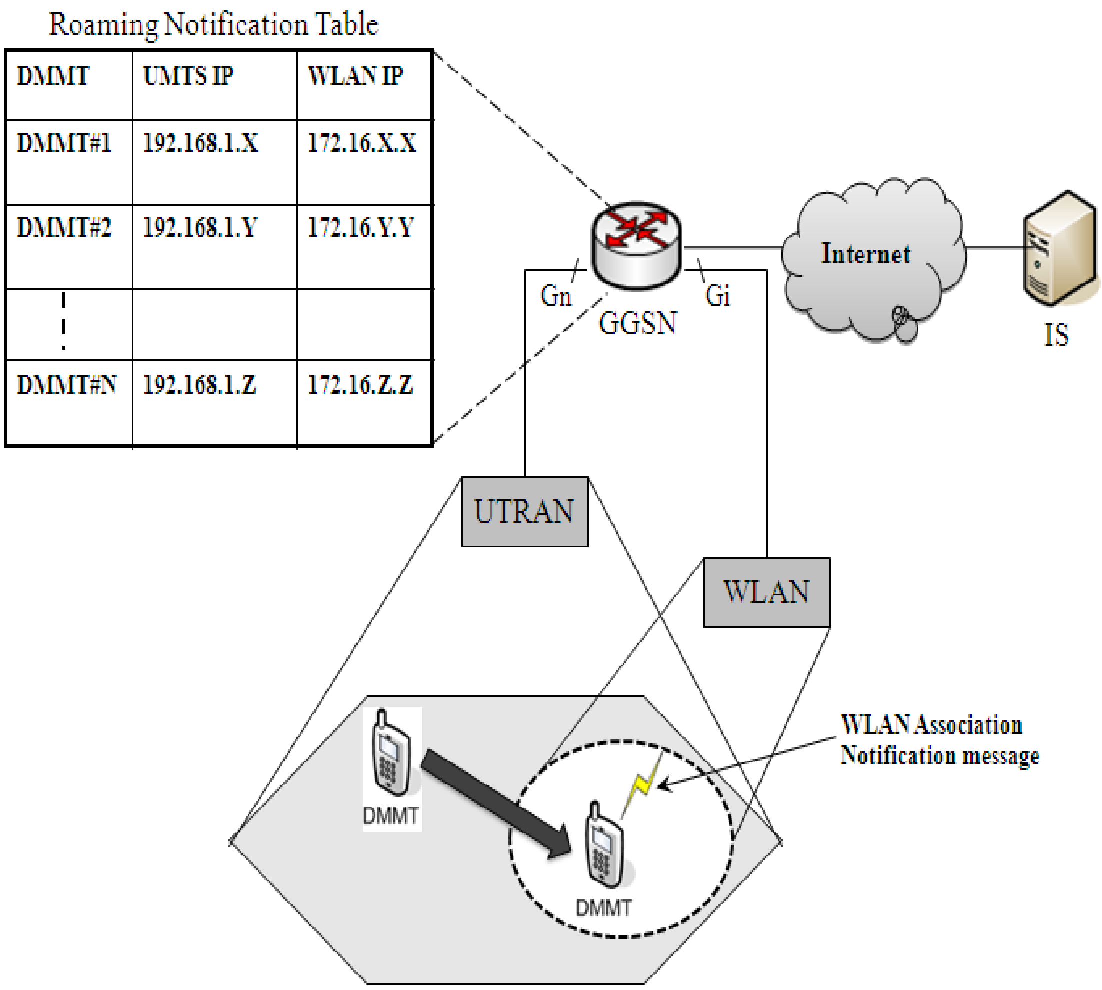

Figure 3.8: Roaming notification table in GGSN

From now on, the CL layer will perform the IP swapping mechanism to mitigate the IP address conflict. The DMMT sends the data packets to the GGSN with the source IP address of the WLAN network. The IP module of the GGSN will allow the data packets to be sent, since, the data packet contains the valid IP address of the WLAN network. Therefore, no ingress filtering issue would appeared. On the basis of destination IP address, the GGSN IP module forwards data packets to the GGSN CL, as illustrated in Figure 3.9. On receipt of data packets, the GGSN CL module will find the matching entry in its RNT. This matching entry would be clearly indicating that the DMMT is moved from the UMTS network and continuing the data session from the WLAN access network. Therefore, the CL module swaps the source IP address field of the data packet by replacing the WLAN IP address with the mapped UMTS IP address before sending the data packets to the PDN. Finally, when the data packets are sent to the IS server, it will have the source UMTS IP address. Therefore, the internet server will be completely transparent with the DMMT movement from UMTS to WLAN network and continue the session without being aware of any modified process.

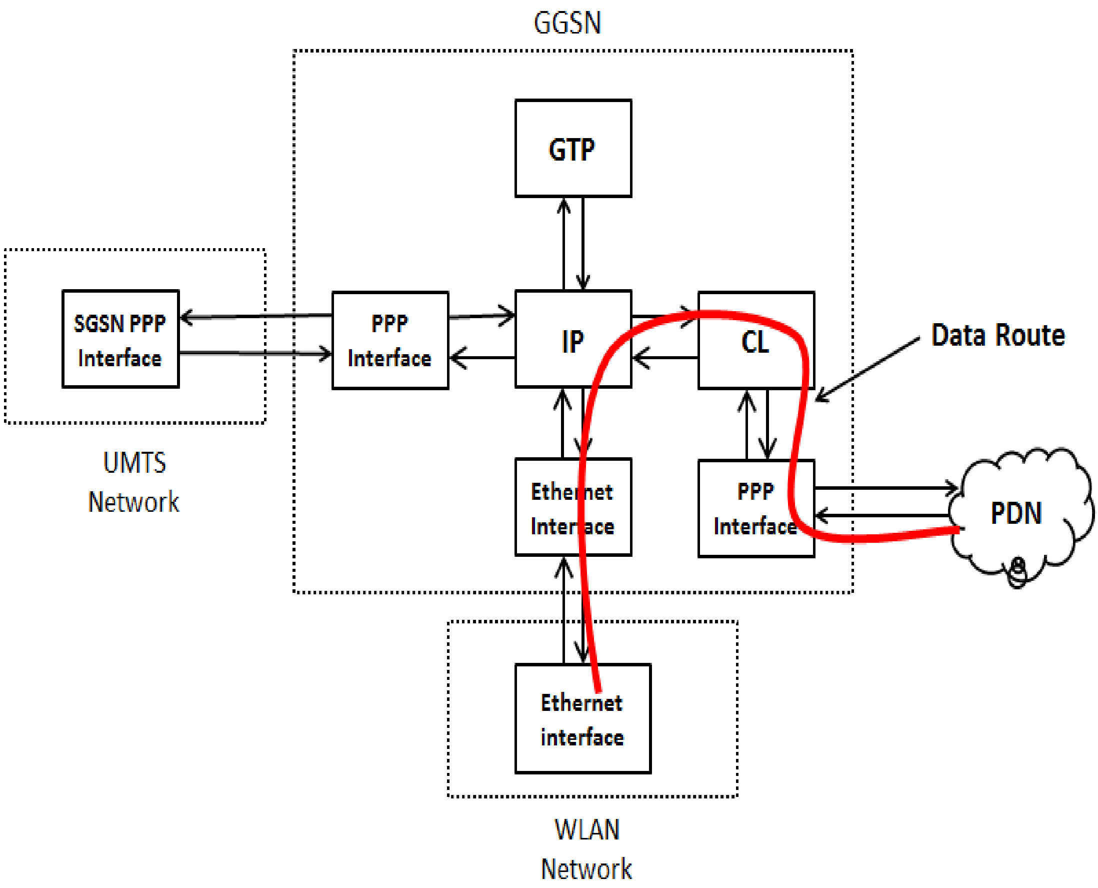

Figure 3.9: Data routing and processing inside the GGSN for WLAN data traffic

In the reverse direction, the IS sends the data packets having the source address field filled with the IS IP address and destination IP address with the UMTS_IP address. On receipt of data packets, the CL will search its database. When the matching entry of UMTS with the WLAN IP address in RNT is found, the CL swaps the destination UMTS IP address with the WLAN IP address and sends it to the IP module. Since, now the destination is the WLAN network, therefore, rather forwarding the data packets to the GTP module for encapsulation, the data packets are sent to the WLAN network. This process of the IP address swap will go on until the DMMT continues its ongoing data session with the WLAN network.

It can be observed that the IP swapping mechanism efficiently addresses the correspondent node’s transparency issue. Moreover, in contrast to the MIPv6 and tight coupling mechanism, the IP swapping mechanism continues the data session in the foreign network without introducing any additional overhead by efficiently reusing the allocated source and destination IP address fields. As described earlier, the CL module is also used in the DMMT for the upper layer transparency. Section 3.5.4 discusses the DMMT CL module operations along with the proposed IP swapping mechanism.

3.5 Design considerations of Dual Mode Mobile Terminal

In Section 3.4, data transportation mechanism inside the UMTS/WLAN network while the wireless client is communicating via UMTS and WLAN access network has been explained. Nevertheless, this section explains the DMMT design and its functionalities to attain the seamless mobility. Before going into the details of design constructions and functionalities of DMMT, it is worth discussing the Multi-homing and cross layer concepts. These functionalities are implemented into the proposed DMMT design for the parallel transmission of the data streams and efficient VHO management.

3.5.1 Multi-Homing Wireless Devices

Nowadays, the wireless devices are equipped with multiple network interfaces to support different wireless technologies, for example, UMTS, WLAN, Bluetooth etc. However, while increasing the options to connect internet via different access networks, seamless mobility cannot be guaranteed when the wireless device is roaming across different wireless access domains. This limitation appears because multiple wireless interfaces cannot be utilized simultaneously for the same data session. Due to the enormous number of Internet Service Providers (ISPs) and the fragmented nature of wireless technologies, the wireless devices must support multiple wireless access networks concurrently to choose the best network for the ongoing data session, among the available wireless access networks. Moreover, if more than one wireless interface can be utilized for the internet connectivity then seamless mobility without any service degradation can be attained.

Therefore, for the mobility management in an integrated wireless heterogeneous network, the concept of multi-homing is very important to be considered which has been ignored by many existing mobility management protocols, for example, MIPv4, MIPv6, SIP etc. Despite many advantages of multi-homing technique, the biggest task in the multi-homing implementation is that how can the correspondent nodes and upper layers of wireless device be made transparent with the multiple IP address assignment to the single data session?

Having identified this issue in the multi-homing feature implementation, the proposed mechanism provides the upper layer and correspondent node's transparency. The correspondent node' transparency has been discussed in Section 3.4.2. Section 3.5.4 explains the upper layer transparency mechanism by using the DMMT CL. In order to implement the multi-homing feature in an integrated UMTS/WLAN network, the SVHOP assigns two different IP address to different wireless interfaces of the wireless device. Each IP address belongs to the particular wireless access network.

As illustrated in the Figure 3.10, the DMMT keeps the UMTS and WLAN IP addresses to communicate through UMTS and WLAN network, respectively. Consequently, internet connectivity can be established via different networks/ISPs. Therefore, not only the best available network can be selected any time for the ongoing session, moreover, an uninterrupted ongoing session switching is also possible because of the make-before-break or soft handoffs realization.

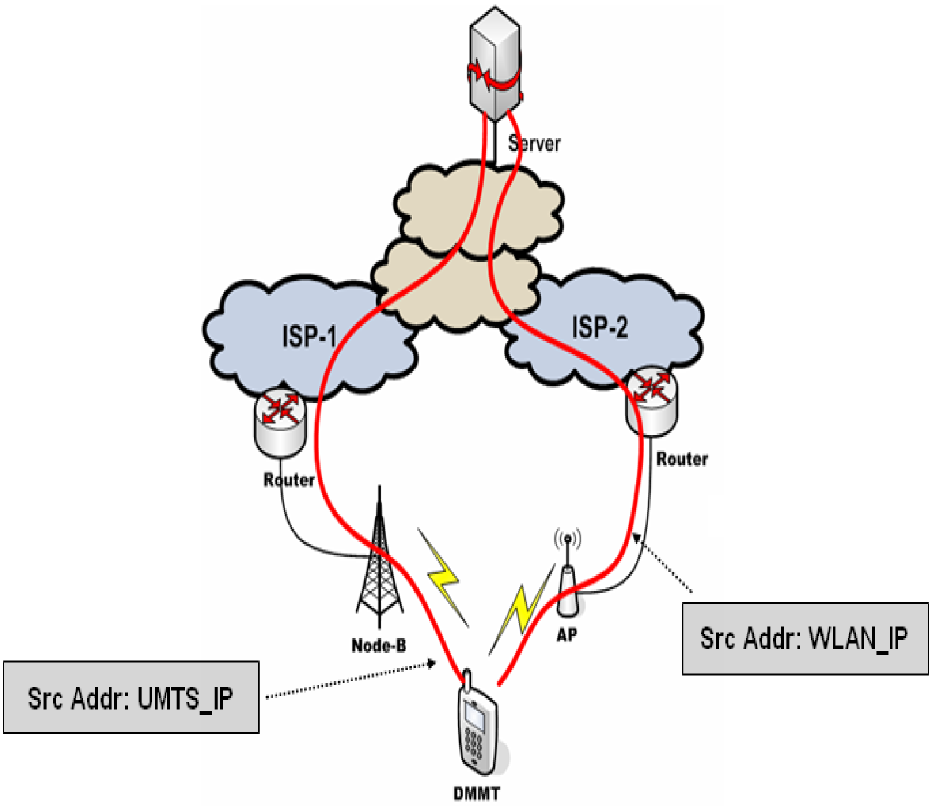

Figure 3.10: Multi-homing DMMT

3.5.1.1 Parallel Transmission

As described earlier, the multi-homing feature enables the parallel transmission over different wireless interfaces of DMMT. Therefore, the handoff latency and packet loss during the network switching can be minimized. To emphasize on the specific parallel transmission mechanism adoption in the proposed mechanism, let us consider three vertical handoff scenarios. Namely these are: without parallel transmission, parallel signaling transmission during handoff and parallel data transmission during and after handoff.

If there is no parallel transmission mechanism implemented then the data session will be highly interrupted during vertical handoff periods. As illustrated in Figure 3.11, during the handoff, when the data are not receiving from interface 1 then the data reception over interface 2 will be followed by the handoff period. This handoff period is consisted of handoff initiation, decision and execution. During this period, no data will be sent and received. Consequently, high handoff latency and packet loss during handoff period will deteriorate the ongoing session.

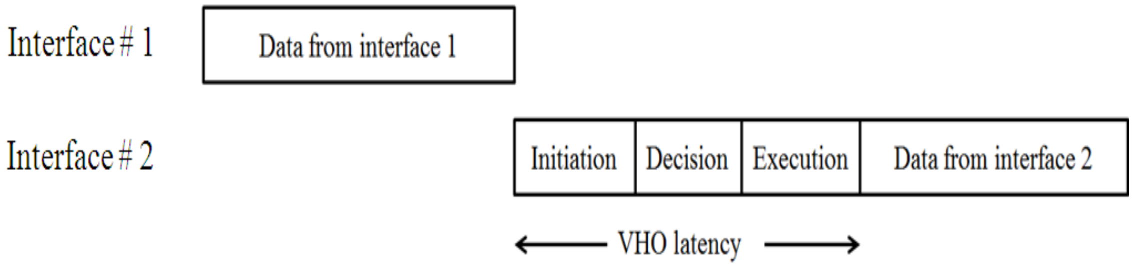

Figure 3.11: Handover mechanism without parallel transmission mechanism

On the other hand, if the mobile device supports parallel signaling transmission then during handoff while the session is establishing with the new access network the data session can be kept alive from the previous network. Therefore, as shown in Figure 3.12, by enabling the make-before-break scenario, handoff latency and packet losses can be minimized.

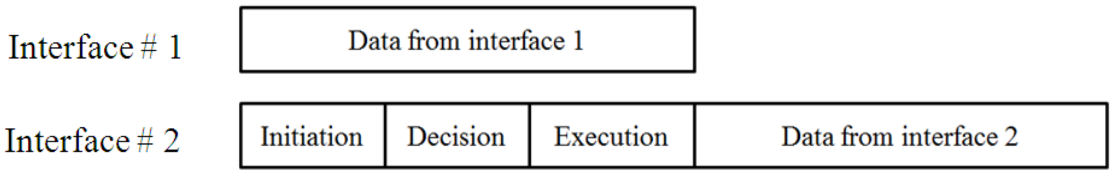

Figure 3.12: Parallel signaling transmission mechanism during handoff

It is worth mentioning here that in the proposed mechanism "parallel signaling transmission during handoff" is implemented. Therefore, after the handoff, the data is sent/received through the new interface only. Section 3.7 and 3.8 explains the signaling flow and performance analysis of the proposed SVHOP mechanism during UMTS to WLAN and WLAN to UMTS vertical handoff scenarios, respectively. The SVHOP ensures minimum handoff latency and packet loss by sending the signaling messages to the target network without losing the active data session with the previous access network.

This mechanism has been opted to avoid situations where both the interfaces will keep sending and receiving the data after the handoff as shown in Figure 3.13. In this case, the scarce wireless bandwidth and network resources will be utilized to send/receive the similar duplicate information which is eventually discarded at the receiver.

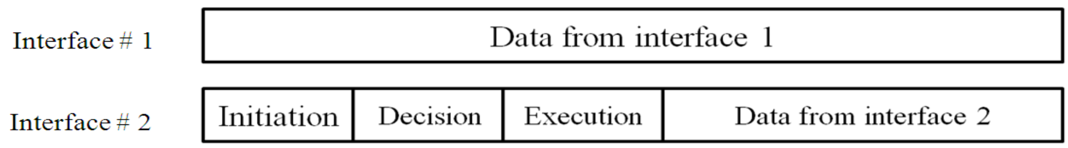

Figure 3.13: Parallel data transmission mechanism during and after handoff

3.5.2 Cross Layer Mechanism

Neither the data link layer nor the network layer VHO mechanism standalone effectively addresses the vertical handoff management. As discussed in chapter 2, the link layer mobility management protocols provide seamless mobility management. Nevertheless, the seamless mobility can be achieved by performing significant modifications in existing network designs and protocol architectures. Such exhaustive alteration increases the network implementation complexity and deployment cost. Consequently, link layer mobility management VHO solutions are not preferred. In contrast, in case of network layer mobility management solutions like MIPV6 provides the network implementation ease. However, when the link layer connectivity with the current access point is broken then the network layer handoff mechanism is triggered to the new access network. Moreover, to continue the session with the new access network, first the link layer is established then layer 3 operations are performed. Such long procedures produce high vertical handoff delays which are not desired in NGNs.

To integrate the strengths of layer 2 and layer 3 integration mechanisms while avoiding the weakness of both, a cross layer VHO protocol (L2+L3) is introduced in this research effort. The proposed SVHOP performs the handoff functionalities over the network layer by receiving the inputs from the data link layer. Unlike the network layer mobility solutions which produce high vertical handoff delay because of the link layer information lacking at layer 3, the proposed mechanism is well aware of the link layer parameters alteration when the MT is roaming across the integrated heterogeneous wireless network. Therefore, rather performing handoff after the existing link is broken with the current network, a proactive VHO approach is adopted in SVHOP. As a result, the VHO latency and packet loss during handoff are minimized. A detail discussion related to the proposed proactive VHO handoff mechanism can be found in the Section 3.6.

In contrast with the link layer mobility management schemes, the SVHOP performs the handoff on the IP layer. For the handoff management the network layer has been chosen because the network layer mobility management protocols provide the independency of access network technologies [53]. Consequently, SVHOP attains the seamless mobility without implementing any major modification in the existing network topologies or protocol architectures.

3.5.3 Construction of the Dual Mode Mobile Terminal

In addition with the integrated UMTS/WLAN network topology, the design of the MT which can support multiple wireless access networks is another challenging task. Since, in the case of integrated networks, a mobile client is required to have UMTS and WLAN interfaces along with the data stream switching mechanism to maintain the session continuity during handover between heterogeneous networks. For example, consider a user is located in the UMTS cell and initiates a session with the internet server with its UMTS interface. When it moves to a region where it receives beacon messages of the WLAN access network; it must switch its active connection to the WLAN to enjoy the high bandwidth and low cost service. Similarly, when it is moving away to the WLAN network coverage, it must switch its active connection to the UMTS to keep its session alive.

Nowadays, the Mobile Nodes (MNs) are equipped with several network interfaces such as WLAN, UMTS, and Bluetooth. Multiple network interfaces in the MT indicates that the data session can be established to any of the available wireless access networks. However, the MT which is able to continue the ongoing session with different access networks is still not realized practically. Therefore, designing of such multiple access networks supportive MT which can support different access networks for the ongoing data session while roaming from one network to another, is considered as another milestone in this thesis.

Since, the WLAN and UMTS network support different data rates, frame structure, modulation schemes, authentication mechanism etc. Therefore, the question arises, how can we design the MT which is capable to switch the ongoing session with different access networks in a seamless manner? To maintain the seamless mobility, we have introduced a wireless device design called as Dual Mode Mobile Terminal (DMMT). Figure 3.14 illustrates the designed architecture of the proposed DMMT protocol stack.

It has been observed that the main difference in WLAN and UMTS workstations exist after the higher layers. Therefore, to have a mobile node that is capable to either connect to UMTS or WLAN networks; here is a need of a mobile node which contains:

- The MAC and Physical layer of both technologies i.e., UMTS and WLAN.
- A mechanism that provides the mobile node intelligence to switch the packets coming from the upper layer of the indented technology interface and efficiently address the handoff operations. For this, we introduced the Convergence Layer (CL).

The CL operates below the IP layer. From UMTS and WLAN network perspectives, this layer operates above the UMTS GMM and WLAN MAC layers, respectively. That is why, we called this CL as 2.5th layer which not only performs the functionalities of routing data packets and dealing with the IP address like a network layer, moreover, it keeps on monitoring the link layer RSS value of the WLAN interface for efficient and proactive network switching decisions.

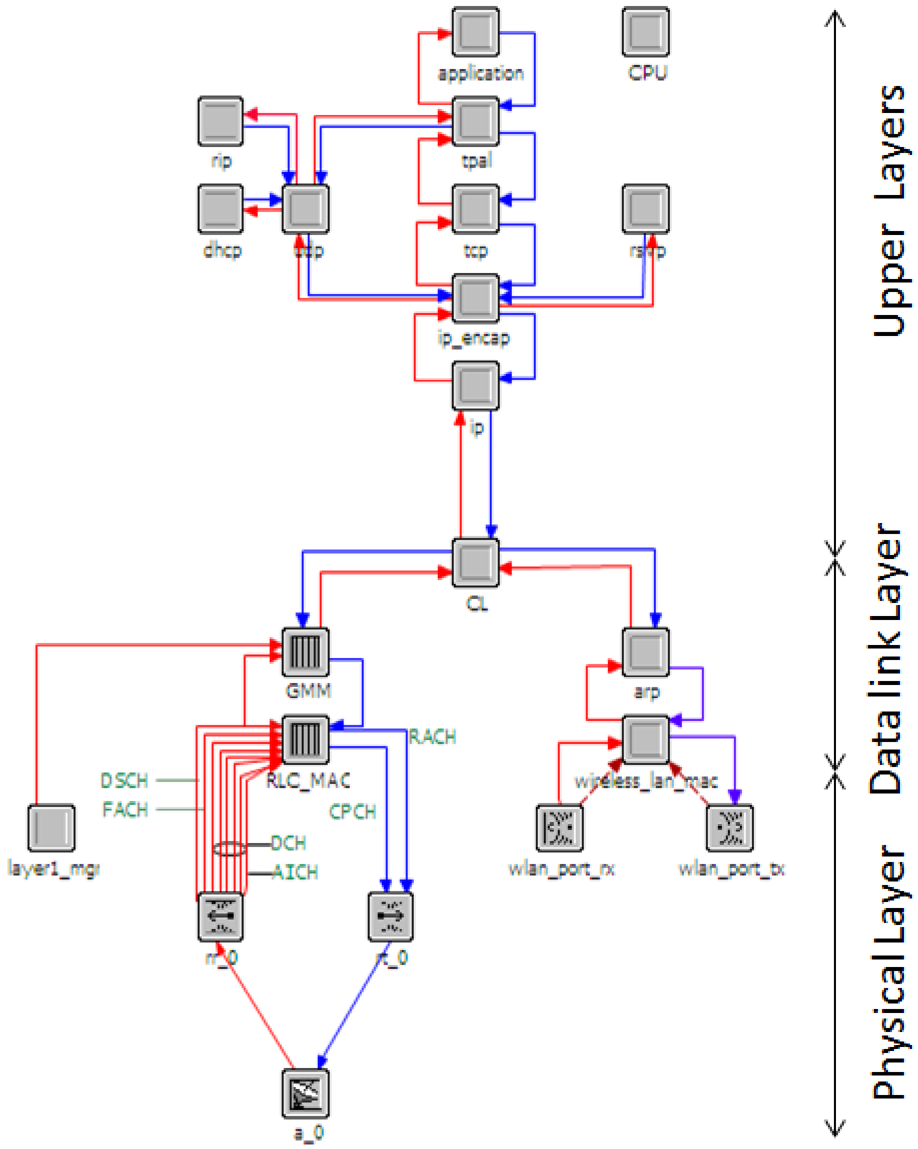

Figure 3.14: DMMT node model in OPNET

3.5.4 Convergence Layer Operations

Essentially, as depicted in Figure 3.15, the CL layer performs the following operation:

- Interface switching for the data packets while the user is moving from one access network to another access network.
- IP address swapping for the upper layers' transparency.

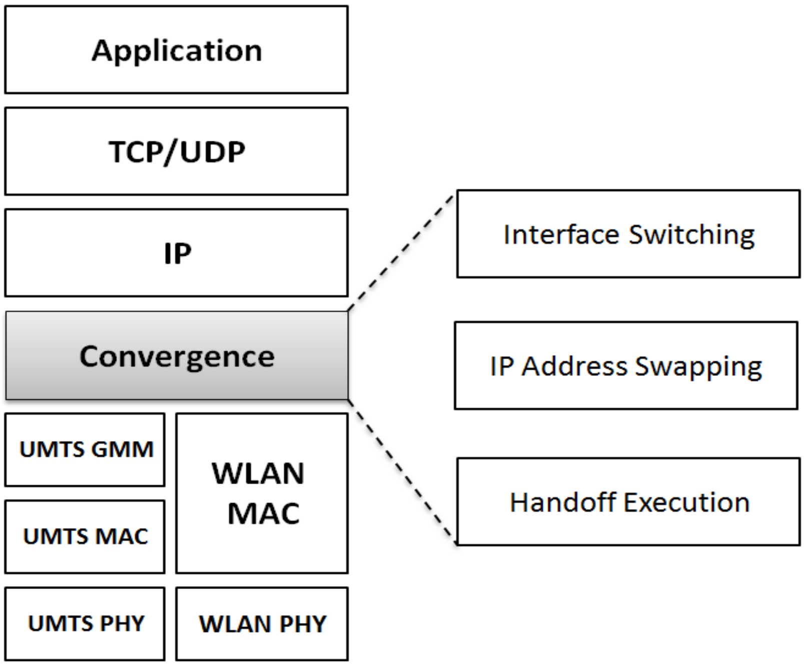

Figure 3.15: DMMT protocol stack and convergence layer features

- Handoff execution on the basis of WLAN link layer RSS hints.

When the user initiates the data session, the selection of the appropriate interface is based on the user's choice. In case of overlay heterogeneous wireless networks, if the user wants the high speed data connectivity and the WLAN network is present then the user may select the WLAN as preferred network. However, if the WLAN network is absent then the UMTS interface will be selected. Nevertheless, while the session is ongoing the network selection/ interface switching must be dynamic, fast and transparent to the user so that uninterrupted data services can be provided.

Figure 3.16 illustrates the detailed flow chart of the CL layer operations. In case of roaming wireless client, the interface switching is performed on the basis of the WLAN_MAC module's "Handoff Messages". Namely, these messages are Switch_WLAN and Swtich_UMTS. Section 3.6.3 explains in detail on what basis the handoff switching messages are transmitted from WLAN_MAC to the DMMT CL layer. When the handoff messages are received, the CL layer performs two functions.

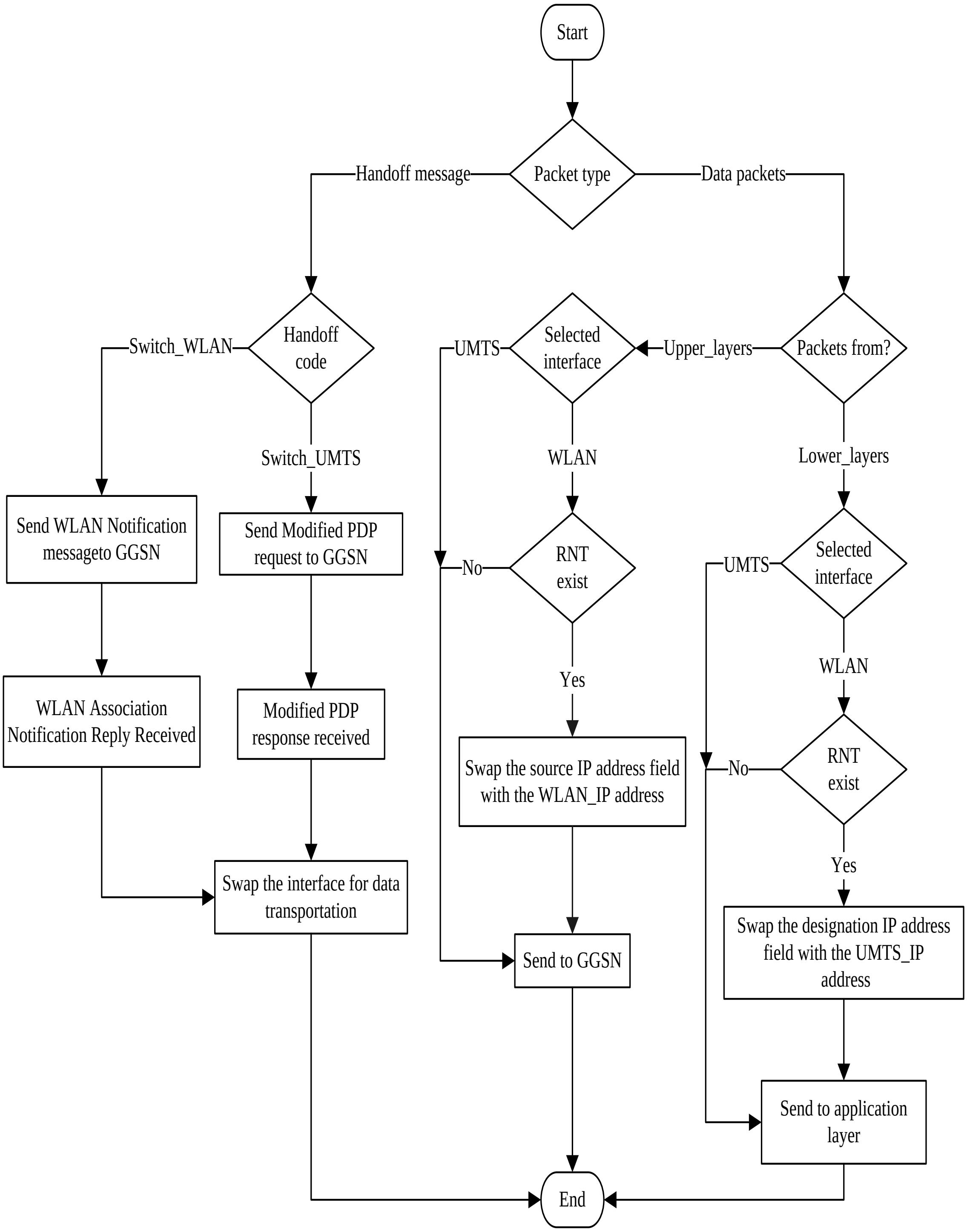

Figure 3.16: Detailed flow chart of DMMT CL layer functions

- Handoff execution: In case of Switch_WLAN message, the handoff is executed by sending and receiving the *WLAN Association Notification* and *WLAN Association Notification reply* messages, respectively, to/from GGSN. However, in case of Switch_UMTS message, the handoff is executed by sending and receiving the Modified PDP request and Modified PDP response messages, respectively, to/from GGSN.

- Interface switching: Once the handoff is executed, the CL layer switches the DMMT link layer interface for the data transportation. This switching can either be from UMTS to WLAN or from WLAN to UMTS. This DMMT interface switching DMMT is termed as interface selection.

In order to understand the operations of different DMMT layers on the data packets, let us assume that the data is coming from application layer. When a specific data stream is transmitted by the application layer, it is broken into segments by the transport layer. Each segment has been assigned a sequence number, so that the data stream can be converted back exactly in the same order at the destination device, as it was transmitted. Later on, these segments are transmitted to the network layer. The Network/IP layer adds the logical addressing and routing information to each segment. When these packets are sent to the convergence layer, it is the responsibility of the convergence layer to send the data packets to the selected link layer interface.

Therefore, if the packets are intended for the UMTS radio access network then the convergence layer routes the data packets to the UMTS GMM layer. The GMM appends the GMM signaling information which includes an International Mobile Subscriber Identity (IMSI), International Mobile Equipment Identity (IMEI) and cell identity, etc. Subsequently, all the packets are sent to the UMTS RLC_MAC. Finally, the packets are sent to the UMTS radio access network via UMTS physical interface of DMMT. However, if the data packets are intended for the WLAN radio access network then the convergence layer routes them to the WLAN MAC. The MAC layer converts the packets into frames and adds a MAC header in each frame that contains the source and destination MAC address information. Consequently, the data streams are sent to the WLAN radio access network via WLAN physical layer of DMMT.

As described in Section 3.4.2, in the proposed mechanism, when the DMMT roams from UMTS to WLAN network, to avoid the ingress filtering it has to acquire and use the new IP address from the current access network to continue its session. At this stage, the session can be sustained by the current IP address, however, the upper layers and the correspondent node recognizes the DMMT session with the pervious IP address i.e., UMTS_IP. The GGSN CL is used to provide the correspondent node's transparency. However, the upper layer's transparency is attained by the DMMT CL layer. Similar to the GGSN CL, the DMMT also contains the RNT table which keeps the UMTS and WLAN IP addresses.

To avoid the IP address conflict by maintaining the upper layer's transparency, the DMMT checks the presence of RNT table when the selected interface is WLAN. When the data packets are arriving from upper layer and the RNT exists then the CL swaps the source IP address field with the WLAN_IP address, before sending the data packets to the GGSN. On the other hand, when the data packets are arriving from lower layer and the RNT exists then the CL swaps the destination IP address field with the UMTS_IP address, before sending the data packets to the application layer. That is how the ingress filtering issue is successfully eliminated. Moreover, upper layer remains transparent to the movement of the DMMT in the foreign network while using different IP address for the same running session.

For the vertical handoff management, a cross layer mobility management mechanism is implemented in the DMMT. The proposed cross layer mechanism enables the DMMT to predict the upcoming VHO events on the basis of link layer RSS hints. Because of the proactive vertical handoff approach, an early handoff is triggered by the DMMT which eventually reduces the handoff latency and packet loss during the handoff periods. In order to predict the upcoming vertical handoff event, a detailed analysis of RSS behavior while moving across the integrated UMTS/WLAN network is conducted. Section 3.6 illustrates the proposed proactive vertical handoff mechanism.

3.6 Evaluation of Proactive VHO Approach

In an integrated UMTS/WLAN internetworking environment, VHOs are performed either to utilize the high bandwidth of WLAN network or due to MN away movement from the WLAN access network. In both cases, the MN should be well-aware and up to date with its current position corresponding to the WLAN Access Point (AP). Therefore, the concept of VHO precision is very important to be considered here.

3.6.1 Lower Layers Awareness

The proposed mechanism enables the DMMT to be lower layers aware. Since, the vertical handoff triggering mechanism is aware with the link layer RSS measurements, therefore, the DMMT can perform its VHO functions in advance for the upcoming VHO event. To understand the significance of the proposed proactive lower layer aware VHO approach, consider the DMMT is moving from WLAN to UMTS network, as illustrated in Figure 3.17.

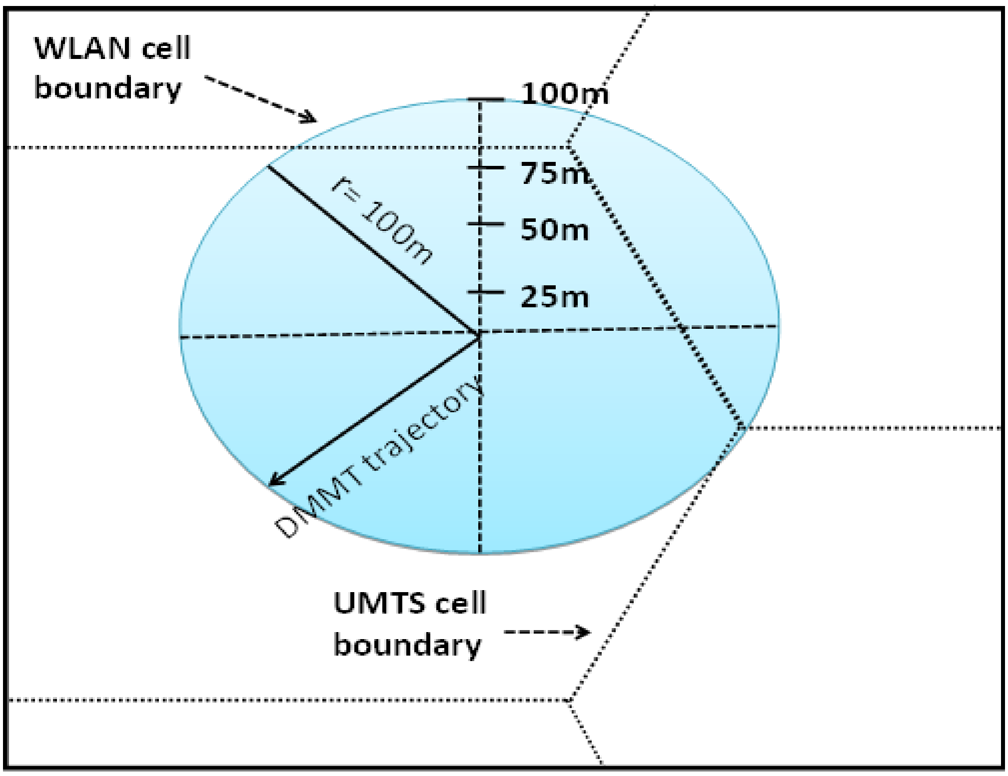

Figure 3.17: DMMT away movements from WLAN AP

When the DMMT starts moving away from the WLAN AP, the RSS value will be decreasing accordingly. When the DMMT reaches to the WLAN boundary, the CL module will be notified by the deteriorated RSS signals. This notification is triggered by the WLAN_MAC module so that the CL layer will execute an early handoff. The early notification of RSS deterioration is essential because if no such an early notification mechanism is present then the DMMT may cross the WLAN coverage region and lost its connectivity with the WLAN AP, even without establishing its connection with the UMTS network. Since, in case of WLAN to UMTS network switching, as shown in Figure 3.17, the situation can relate to a non-overlapping region and a break-before-make event may take place. Consequently, an abrupt service disruption or a call break event can be experienced. Therefore, to avoid such events and to adopt a make-before-break mechanism, a proactive vertical handoff approach is designed and implemented. Nevertheless, to avoid the wrong VHO initiation and execution, precise link layer measurements are required.

3.6.2 RSS Measurements Based on Log Normal Shadowing Model

The precise RSS values answer two crucial questions. First, whether a handoff is required? Second, when and to which network the handoff should be triggered? In other words, when the DMMT is entering the WLAN or when it is exiting the WLAN network?

To answer the aforementioned questions, we have to look deep inside the RSS behavior in a propagation channel and the parameters that influence the RSS. The RSS has a direct relationship with the path loss of radio signal. The path loss is the phenomenon which occurred when the signal travels into the propagation medium. The path loss increases/decreases with the increasing/decreasing distance between the transmitter and receiver. Moreover, the path loss is also influenced by several other factors such as the environment (rural or urban, foliage and vegetation), terrain type, and propagation medium (moist or dry air).

In order to predict the path loss behavior of radio waves during propagation, radio propagation models are used. The radio propagation models are empirical mathematical formulae to describe the predicted performance of radio waves in propagation medium. The study of the radio propagation model describes the effects of medium constraints on the path loss values. In addition, it helps to determine the network coverage.

To provide the intelligence to the Dual Mode Mobile Terminal (DMMT) to make the handoff decision accurately, we evaluated and implemented Log Normal Shadowing Model (LNSM). By using the LNSM, precise calculation of path loss and RSS with the corresponding DMMT position with the WLAN AP was measured, while the DMMT is roaming across the integrated wireless access networks.

The log-normal shadowing propagation model can mathematically be expressed as [86]:

$$PLoss = L + \eta * 10_{log}\,(d) + X_{\alpha} \tag{3.3}$$

Where, *PLoss* represents the path loss in dB, *L* is the constant power loss in propagation medium, the parameter η is a path loss index and its value depends upon the propagation environment (generally in between 2 to 4 [86]), larger the η means more obstacles in the propagation environment; d is the distance between the transmitter and receiver. The parameter Xσ represents shadowing fading with zero mean Gaussian random variable and standard deviation σ with values between 6-12 dB depending on the environment. The RSS is expressed as [86]:

$$P_r = P_t - P_{Loss} \tag{3.4}$$

Where, *Pr* represents the received power in dBm and *Pt* is the AP transmitted power in dBm. It should be noted that the default power transmitted by the WLAN APs is 100mW (20dBm) [87].

In order to study the RSS behavior in the propagation channel, a simulation scenario has been created in which initially the DMMT is placed near to the WLAN AP. However, as the simulation time progresses the DMMT start moving away to the WLAN AP. The values of propagation distance along with the corresponding path loss and received power measurements were collected. Figure 3.18 and Figure 3.19 are demonstrating the relationship of path loss and RSS with the propagation distance, respectively. It can be observed that the path loss exponentially increases; conversely, the RSS exponentially decreases when the distance between the AP and DMMT increases.

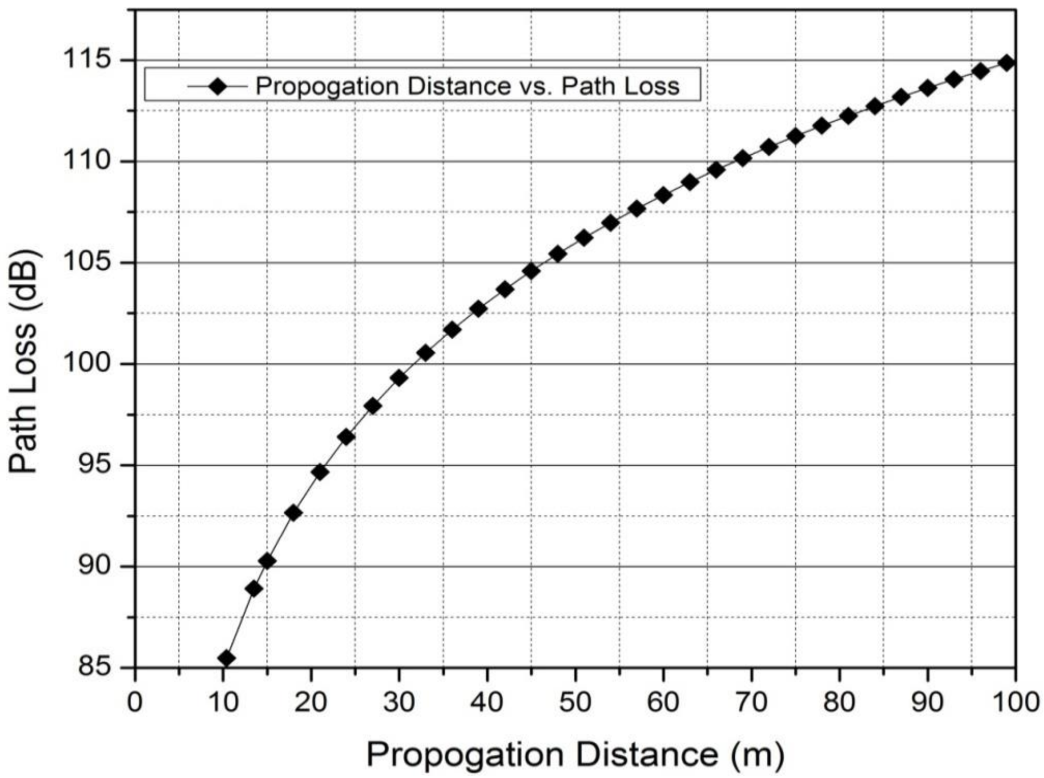

Figure 3.18: Relationship of path loss with propagation distance

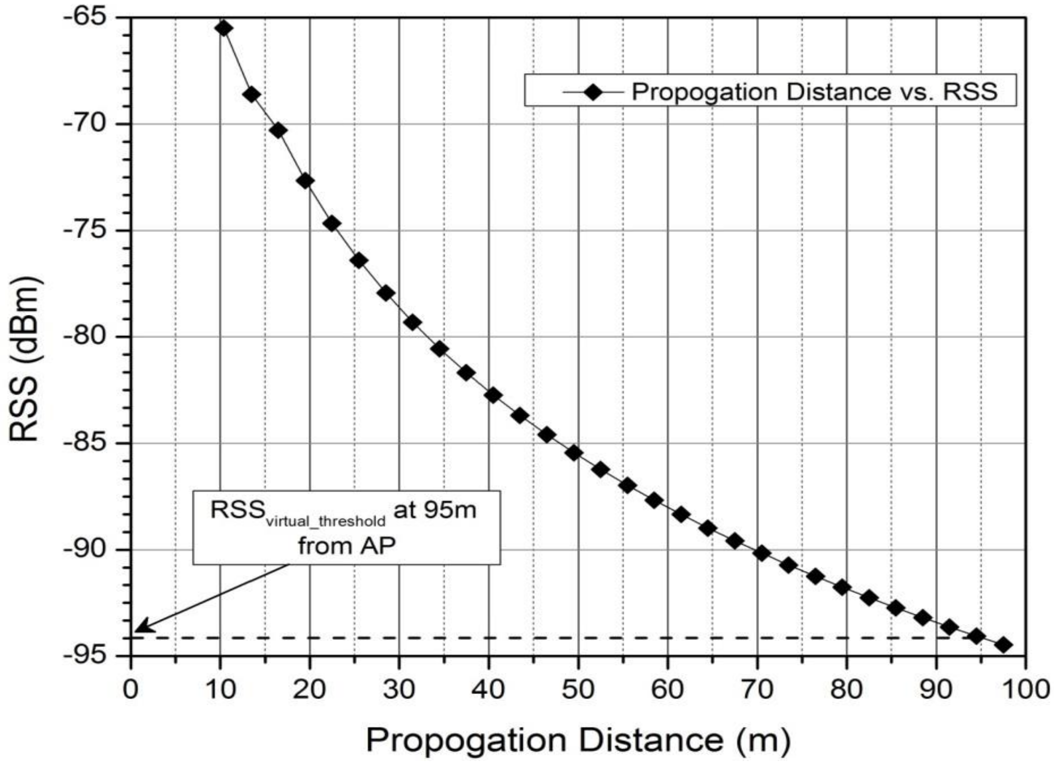

Figure 3.19: Relationship of RSS with propagation distance

It is apparent from Figure 3.19, at 100 meters away from the AP, the DMMT reaches to the RSS threshold (RSSψ) which is set as -95dBm. Since, below RSSψ, here is no guarantee of correct reception of data packets by the DMMT. Therefore, by using the LNSM on the applied parameters, the coverage area of WLAN cannot be extended more than 100 meters.

3.6.3 Handoff Decision and Execution by Using RSS Measurements

From Section 3.6.2, we can observe that the DMMT is unable to properly receive the transmitted signals from WLAN AP when it is 100m away from the AP. According to the legacy techniques of VHO mechanism, after 100m away from the AP the DMMT has to initiate an upward VHO mechanism. Since, there was no mechanism to sense the DMMT movement; consequently, after losing the current session the DMMT invokes the VHO mechanism to the UMTS network. In other words, the link layer connection to the WLAN network is broken first then the VHO mechanism was executed. Nevertheless, such reactive VHO approach increases the VHO latency and packet loss.

In order to minimize the VHO latency and packet loss in an integrated UMTS/WLAN internetworking environment, the proposed mechanism provides the intelligence to the DMMT to be aware of the upcoming VHO event. The proactive approach results early VHO initiation and execution. Consequently, the low latency handoff along with the low number of packet loss can be achieved.

Now the question is when the DMMT will realize that it is moving away from WLAN AP? And when it is supposed to execute the VHO to UMTS network? From the statistics collected from the above DMMT moving away from the WLAN AP scenario, for the upward vertical handoff case, it is suggested that the DMMT should perform upward VHO when its RSS reaches to 99% of RSSψ (we say it virtual RSSψ (RSSv̈ψ)).

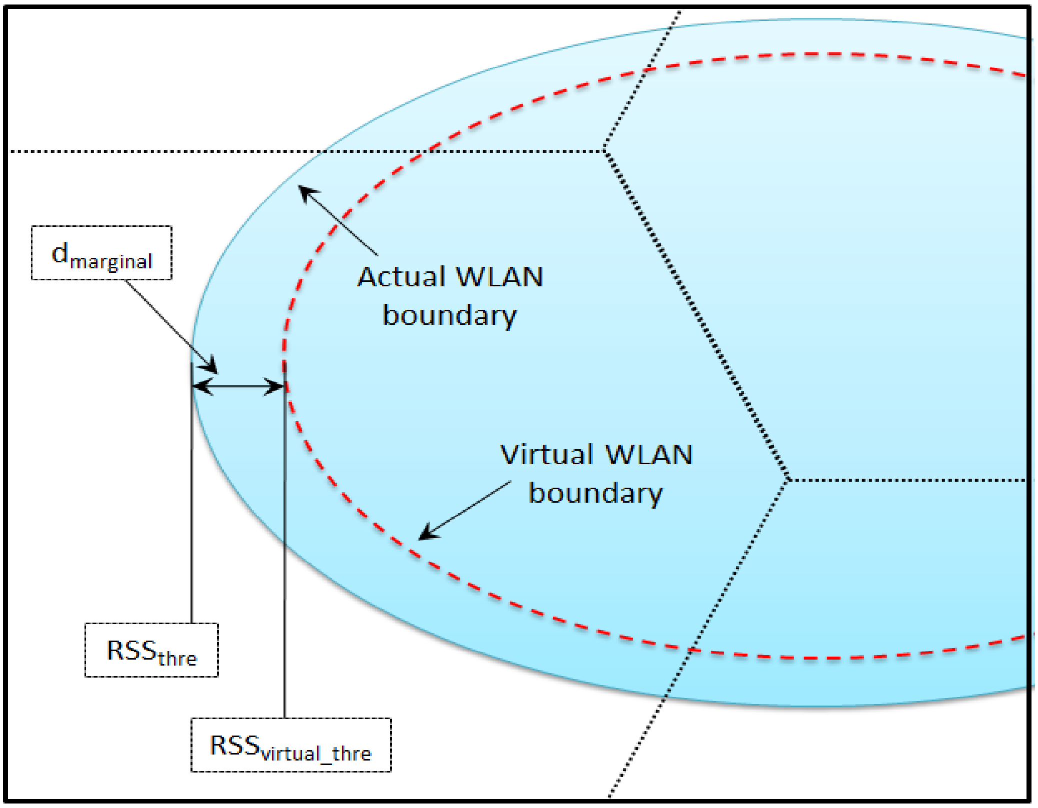

Figure 3.20: Creation of virtual WLAN cell boundary

This value is selected because if we closely observe the RSS values in Figure 3.19, it can be find out that at RSSv̏ψ; the DMMT is approximately 95 meters away from the AP. If the handoff is initiated and executed at this RSSv̏ψ, then the DMMT would have the margin of approximately 5m to perform the handoff before going out of coverage. Therefore, in this marginal region, which is termed as $d_{marginal}$, the DMMT would be triggering the VHO with the UMTS network meanwhile collecting the PDUs from WLAN network. As illustrated in Figure 3.20, by implementing this proactive approach a virtual WLAN boundary is created for the DMMT at 95 meters away from the WLAN AP where the RSS is about 99% of RSSψ.

This margin is suitable enough to complete the VHO towards the UMTS network while remain connected to the WLAN network. If this virtual boundary is very close to the actual WLAN boundary then there is a chance that the DMMT can move out of coverage before establishing the connection with the UMTS network. On the other hand, if the marginal region between the virtual and actual WLAN boundary is wide then the DMMT would switch its active connection to the UMTS from the WLAN network, even located deep inside the WLAN region. Consequently, an early handoff will result in attainment of the high cost and low data rate services of the UMTS network.

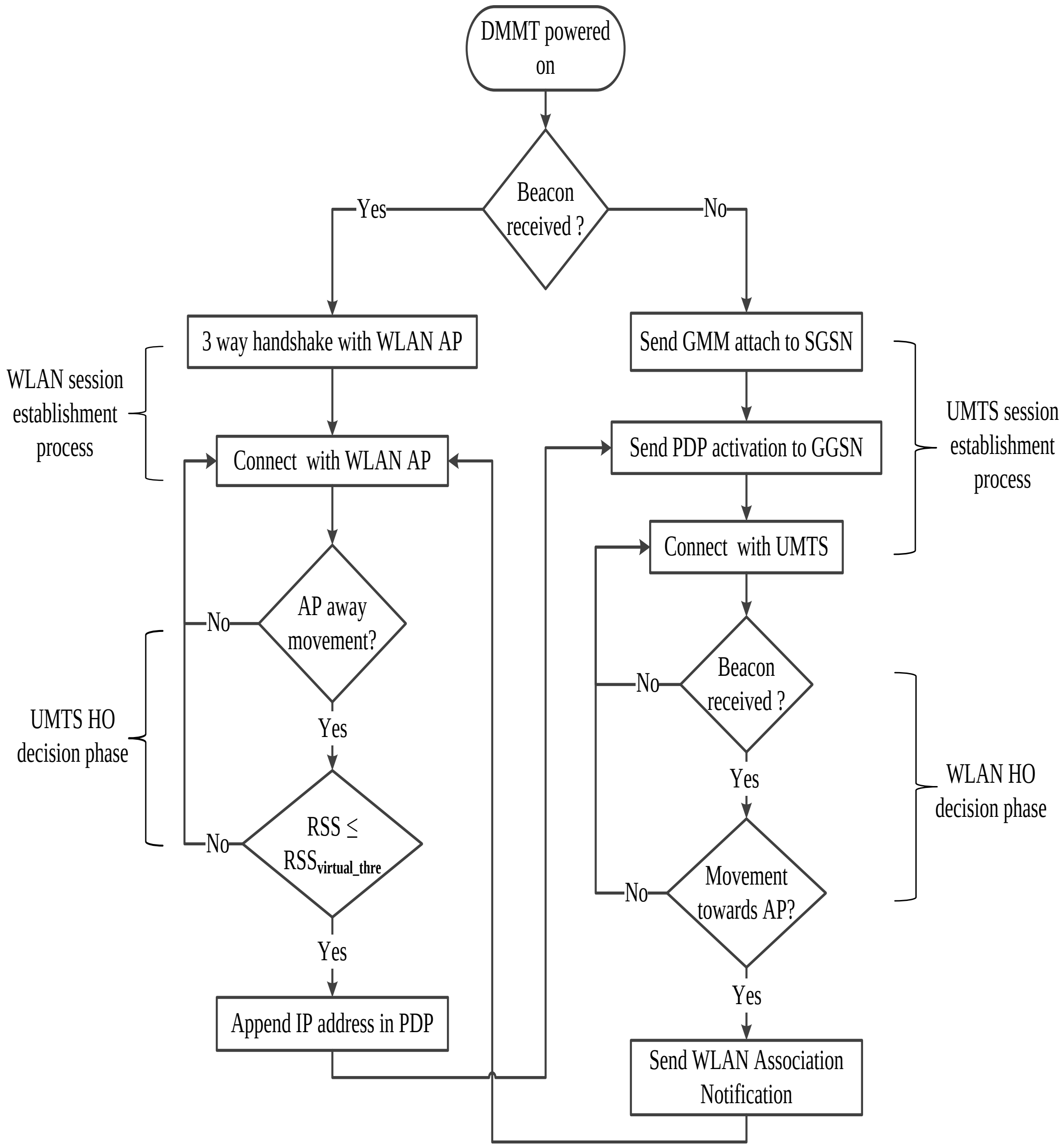

Figure 3.21: Flow chart of DMMT powered on and upward/downward VHO scenario

Figure 3.21 illustrates the basic flow chart of DMMT powered on and upward/downward VHO scenario in an integrated UMTS/WLAN internetworking environment. When the DMMT is powered on, it searches for the WLAN beacon messages. If no beacon messages are received by the DMMT then it will be connected with the UMTS network. To connect with the UMTS network, the DMMT has to perform GMM attach and PDP context activation with the SGSN and GGSN, respectively. The DMMT moves in the integrated UMTS/WLAN network with the velocity of 1.5m/s (pedestrian user speed). The DMMT WLAN interface continuously scans the air interface for the possible existence of the WLAN network. At any point, when the WLAN MAC receives the beacon messages, it will suggest the DMMT CL module when and to which network the handoff should be triggered. This handoff suggestion is sent to the DMMT CL on the basis of DMMT WLAN MAC module's upward and downward vertical handoff decision algorithm.

Figure 3.22 represents that the UMTS to WLAN and WLAN to UMTS handoff decision is made on the basis of RSS measurements. For example, the consecutive increasing strength of RSS indicates that the DMMT is moving towards the WLAN AP. Similarly, the consecutive decreasing RSS strength is a sign that the DMMT is moving away from WLAN AP. However, it is worthwhile mentioning here that in case of UMTS to WLAN VHO, it is preferred to perform the VHO as soon as the DMMT moves inside the WLAN coverage region so that the high data rate and low cost data services can be attained. In contrast, in case of WLAN to UMTS VHO, it is preferred that even the RSS measurement trend shows that the DMMT is moving outside the WLAN, the DMMT should keep the WLAN connection until it reaches to the WLAN network virtual boundary. That is how the low cost and high data rate services of the WLAN can be kept as long as possible.

Once the decision is made that the DMMT is moving towards WLAN AP then the WLAN MAC module sends Swith_WLAN message to CL. On receipt, the CL executes the downward VHO by sending the *WLAN Association Notification* message to the GGSN.

Downward and upward VHO decision algorithm (Beacon messages)

```
Let, MAX = 5, Arr [MAX], downward_trend = 0, upward_trend = 0, WLAN_ selected =
false.
Step1. Let Receive Signal Strength be r, {r: r ∈ RSSn},∀n,
if (r from AP) then
    goto Step 2
end if
else
    Connect to UMTS network
end if
Step2. DMMT comes in a coverage region of an AP
if (r ≥ rψ) then
    Insert current value of "r" into current location of Arr [ ]
Step3.  Interface selection
    If (current index of array == MAX) then
        for (m = 0; m < MAX; m++) do
            if ((Arr [r_m] < Arr [r_m+1]) && WLAN_ selected = false) then
                downward_trend ++;
                if (downward_trend == MAX) then
                    Send Switch_WLAN message to CL
                    WLAN_selected = true
                end if
            end if
            else if ((Arr [r_m] > Arr [r_m+1]) && WLAN_ selected = true) then
                upward_trend++;
                if (upward_trend == MAX) then
                    if (r ≤ r v̈ψ) then
                        Send Switch_UMTS message to CL
                        WLAN_ selected = false
                    end if
                end if
            end if
        end for
    end if
end if
```

Figure 3.22: Pseudo code of the upward and downward VHO decision algorithm based on the L2 RSS measurements

To perform the upward handoff, we designed the DMMT propagation trajectory in a manner that after the downward VHO, the DMMT started moving back to the UMTS network. Because of the away DMMT movement from AP, the RSS will constantly be deteriorating in accordance with the increasing distance between DMMT and AP. Hence, the decreasing RSS trend would be a clear indication that the DMMT is moving away to the WLAN AP. At this stage, the DMMT will wait the RSS to become equal or lesser than RSSv̈ψ to execute handoff.

When handoff condition meets, the CL appends the IP address in the PDP message and sends to the GGSN. In general, the PDP field is empty and to be filled by the GGSN during the PDP context activation process. Therefore, an already filled IP address field in the PDP context request message (which is termed as Modified PDP context request message) notifies the GGSN that this is basically a handoff request from WLAN to UMTS network. The GGSN then searches its database to find the sent IP address. When the GGSN finds a matching IP address entry in its RNT, it starts routing the data packets to the UTRAN. Henceforth, all upcoming data packets will be received by the DMMT at its UMTS MAC interface.

In case of downward and upward VHO execution requests from the DMMT to the GGSN, the GGSN completes the HO procedure by updating the RNT status and routes the data packets on the requested network. The RNT matching IP address entry is created in case of downward VHO request, whereas, the RNT matching entry is deleted in case of upward VHO execution request. The DMMT do not receive the data packets on its new interface until the handoff process is completed [88] and the GGSN routes the data packets at DMMT new attachment point.

3.7 Signaling Flow of Network Nodes during VHO

This section discusses in detail the step by step procedure of the UMTS connection establishment, WLAN connection establishment, UMTS to WLAN handoff, and WLAN to UMTS handoff procedures along with all the corresponding signaling messages.

3.7.1 UMTS Connection Establishment

It is assumed that initially DMMT is located under the coverage of the UMTS access network. In order to access the internet servers, DMMT has to perform the GMM attach and PDP context procedures. The GMM attach procedure is a three way handshake between the DMMT and SGSN. Without the GMM attach procedure, DMMT is not reachable by the network because SGSN does not have the routing or location information of DMMT. This procedure makes the DMMT known to the network and SGSN collects the security, location management and access control parameters input from the DMMT. Figure 3.23 illustrates the three-way handshake between the DMMT and SGSN to perform the GMM attach procedure. Over the Radio Resource Control (RRC) protocol, the DMMT sends the *GMM: Attach Request* to the RNC. This request is further forwarded to the SGSN over Radio Access Network Application Part (RANAP) protocol. The *GMM: Attach Request* contains the IMSI, IMEI and location information of the DMMT. Upon receipt of the *GMM: Attach Request*, SGSN searches the provided entries in its database. When the provided entries are verified, the SGSN sends back the *GMM: Attach Accept* message to the DMMT. In response, the DMMT sends the *GMM: Attach Complete* message to complete the attach procedure. Consequently, now the DMMT is authorized in the UMTS network and the PS signaling connection is established between DMMT with the UMTS core network.

In order to initiate the data session with the internet servers, i.e., external PDN, the DMMT has to perform the PDP context procedure. Actually, after establishing a GMM attach procedure, the DMMT is only known to the SGSN node. However, to communicate with the external packet networks, DMMT has to initiate the PDP context. This internetworking with the external PDN is arranged by the GGSN. The GGSN keeps the routing information of PS-attached DMMT. The GGSN creates the tunnel to send the data to the DMMT to its current point of attachment.

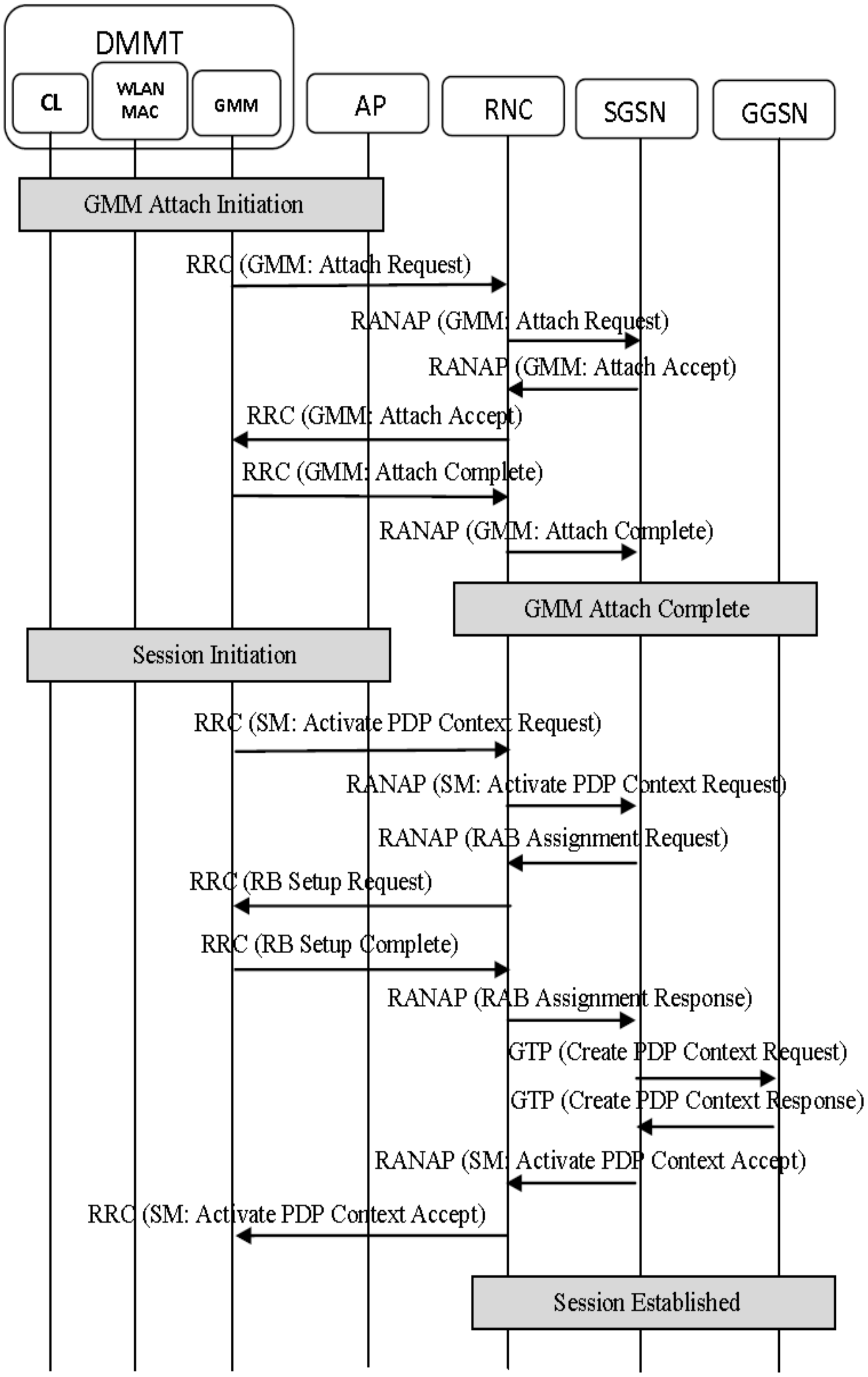

Figure 3.23: UMTS connection establishment

To create a PDP connection, the DMMT sends a *SM: Activate PDP Context Request* to the SGSN via RNC. This message contains several fields such as PDP address, Network Service Access Point Identifier (NSAPI), QoS and Access Point Name (APN). The PDP address or an IP address is requested by the DMMT that will be used for the PDN communication. The IP address field remains empty and to be filled dynamically by the GGSN. The NSAPI is used to identify the PDP context. For different PDP context of DMMT different NSAPI values is used.

The DMMT can have multiple simultaneous PS calls at the same time. For this, multiple PDP contexts with different QoS can be activated. Multiple PDP contexts can be divided into two categories.

- Multiple primary PDP contexts: are used to access different PDNs. Every primary PDP context has a different IP address.

- Multiple secondary PDP contexts: are used when the primary PDP context is already activated.

By the secondary PDP contexts, multiple connections to the same PDN can be triggered with different QoS. Every secondary PDP context has the same primary PDP context IP address. For simplicity, a single PDP context is illustrated in Figure 3.23. However, whenever required multiple primary and secondary PDP contexts can be activated any time. The APN is an optional parameter which is basically the GGSN address.

On receipt of *SM: Activate PDP Context Request*, the SGSN begins the Radio Access Bearer (RAB) association and sends a *RANAP: RAB Assignment Request* to the RNC. This begins the RAB allocation for the required QoS with the unique RAB ID. The RAB is a part of PDP context activation that operates between DMMT and RNC. More specifically, it is a service that operates on the access stratum to support the non-access stratum and provides data services from DMMT to the core network.

On receipt of *RANAP: RAB Assignment Request*, the RNC inspects the availability of the downlink and uplink capacity. If the RNC can facilitate the DMMT with the requested QoS, then it sends a *RRC: RB Setup Request* to the DMMT to establish an appropriate Radio Bearer (RB). If the requested QoS is not available at the RNC, then the RNC will send the *RRC: RB Setup Request* with the negotiated QoS parameters. The DMMT establishes the assigned RB as requested by the RNC and send back a *RRC: RB Setup Complete* message. Finally, *RANAP: RAB Assignment Response* is sent to the SGSN as a RAB-RB mapping notification.

After that the SGSN sends a *GTP: Create PDP Context Request* message with the negotiated QoS value to the GGSN and request it to create a tunnel between them. The Tunnel Endpoint IDentifier (TEID) is enclosed in this message. After the tunnel establishment, the GGSN dynamically assigns an IP address to the DMMT and sends it to SGSN in *GTP: Create PDP Context Response*. On receipt, SGSN sends a *RANAP: Activate PDP Context Accept* message to the RNC. Finally, RNC sends an IP address to the DMMT by sending the *RRC: Activate PDP Context Accept* message. Consequently, a session is established and the DMMT is ready to send the Packet Data Units (PDUs) to the targeted PDN with the requested/negotiated QoS parameters.

3.7.2 UMTS to WLAN Switching

After the successful establishment of UMTS data session, the DMMT starts moving towards the WLAN AP coverage region. Even the data session is ongoing via UMTS network; the DMMT continuously scans the nearby wireless networks for the possibility of connection establishment with other high speed wireless network. In this case, it is WLAN. As illustrated in Figure 3.24, when the WLAN_MAC module of DMMT starts receiving the beacon frames from the WLAN AP, a three way handshake procedure is taken place between the WLAN_MAC and WLAN AP for the association with the WLAN. The WLAN_MAC sends an *Association Request* message to the WLAN AP, which is replied back with the *Association Response*. Finally, the WLAN_MAC sends an *Association Acknowledge* message to WLAN AP and then DMMT connects to the WLAN network. Meanwhile the DMMT establishes its L2 connection with the WLAN network; the WLAN MAC module will start computing the RSS measurements in parallel. When the RSS trend will ensure that the

DMMT is moving towards WLAN network, the decision to switch the active data with the WLAN network would be finalized. Now, the DMMT MAC module sends Switch_WLAN message to the DMMT CL module.

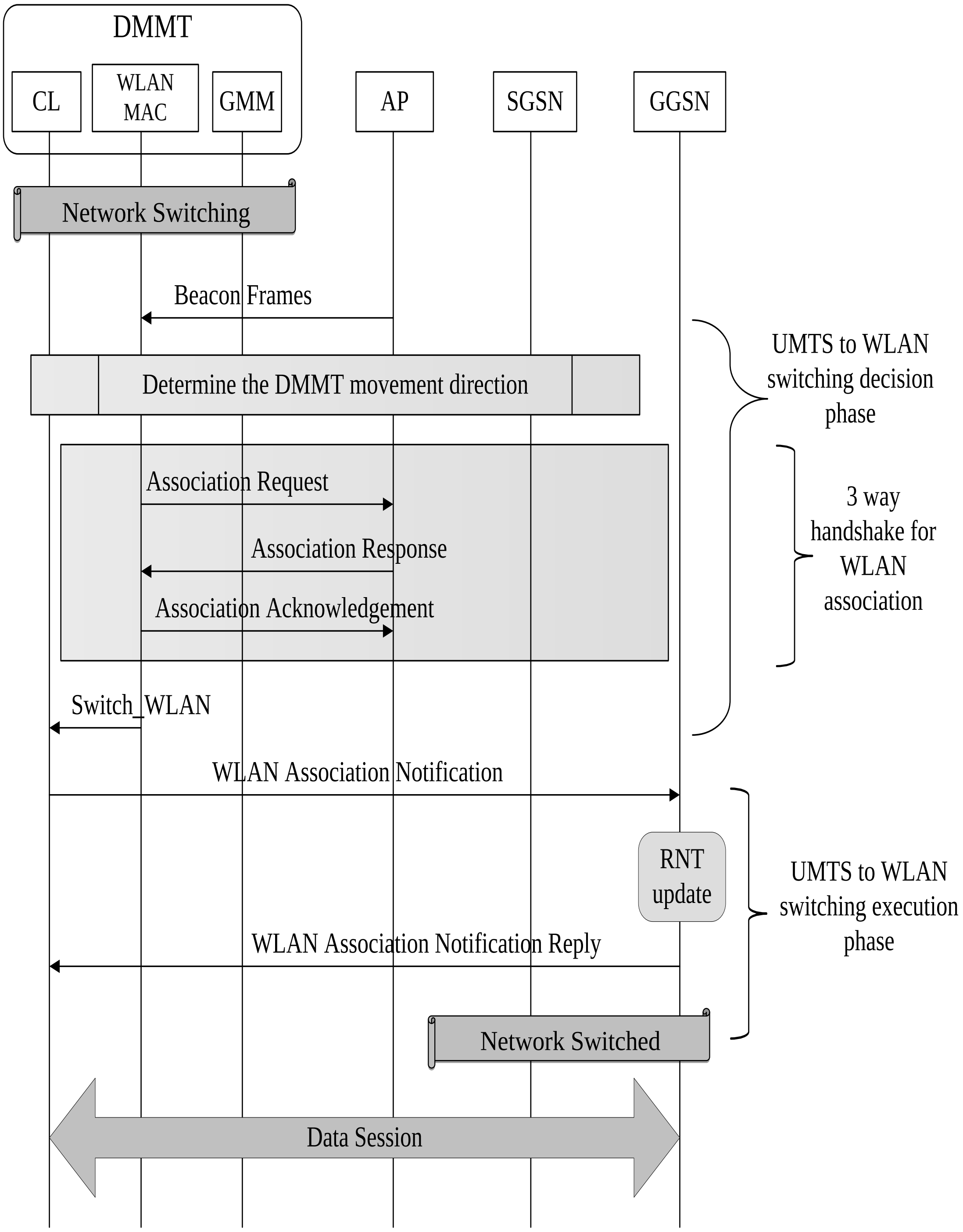

Figure 3.24: SVHOP for UMTS to WLAN switching

At this stage, the DMMT has to continue the active data session with the WLAN IP address. However, to route the internet servers PDUs not from the UTRAN but from the WLAN network, the DMMT will have to notify the GGSN about its relocation. For this, the DMMT CL module sends the *WLAN Association Notification* message to the GGSN. As described in Section 3.4.2, this notification message contains the WLAN and UMTS IP addresses. Actually, by this double entry IP address message the GGSN will be notified that this is the same terminal, who initiated the session in the UTRAN with the dynamically assigned UMTS IP address, now moving to the WLAN with the notified IP address. On receipt of this notification, the GGSN will keep this entry in the RNT. This relocation process is completed when the GGSN sends the WLAN *Association Notification Reply* message to the DMMT CL. Henceforth, all the PDUs for the DMMT UMTS IP address will be routed to the mapped WLAN IP. Therefore, instead of sending the PDUs towards SGSN via Gn interface, the GGSN will start sending the PDUs on its Gi interface, which is connected to the WLAN network.

3.7.3 WLAN to UMTS Switching

When the DMMT starts moving away from the WLAN AP, the RSS value will be decreasing accordingly. These regular RSS dropping values are noted by the WLAN_MAC module so that the CL module will be informed for the proactive VHO. This early VHO triggering mechanism is adopted because the RSS deterioration represents the DMMT away movement from WLAN network. If no such an early notification mechanism is present then DMMT may cross the WLAN coverage region and lost its connectivity with the WLAN AP, even without establishing its connectivity with the UMTS network. Since, in case of WLAN to UMTS network switching, situation can relate to a non-overlapping region and a break-before-make event will take place, consequently, an abrupt service disruption or a call break event can be experienced. When CL is notified by the deteriorated RSS signals by sending the Switch_UMTS message then instead of establishing UMTS connectivity after losing WLAN connectivity, an early execution of VHO will be made by the CL. As a result of this proactive approach, a PDP context request will be sent to UMTS network by the GMM module when:

$$RSS \leq 99\%\ RSS_{Thershold} \qquad (3.5)$$

Consequently, as shown in Figure 3.25, before breaking the active connection with the WLAN network, the DMMT will start establishing its session with the

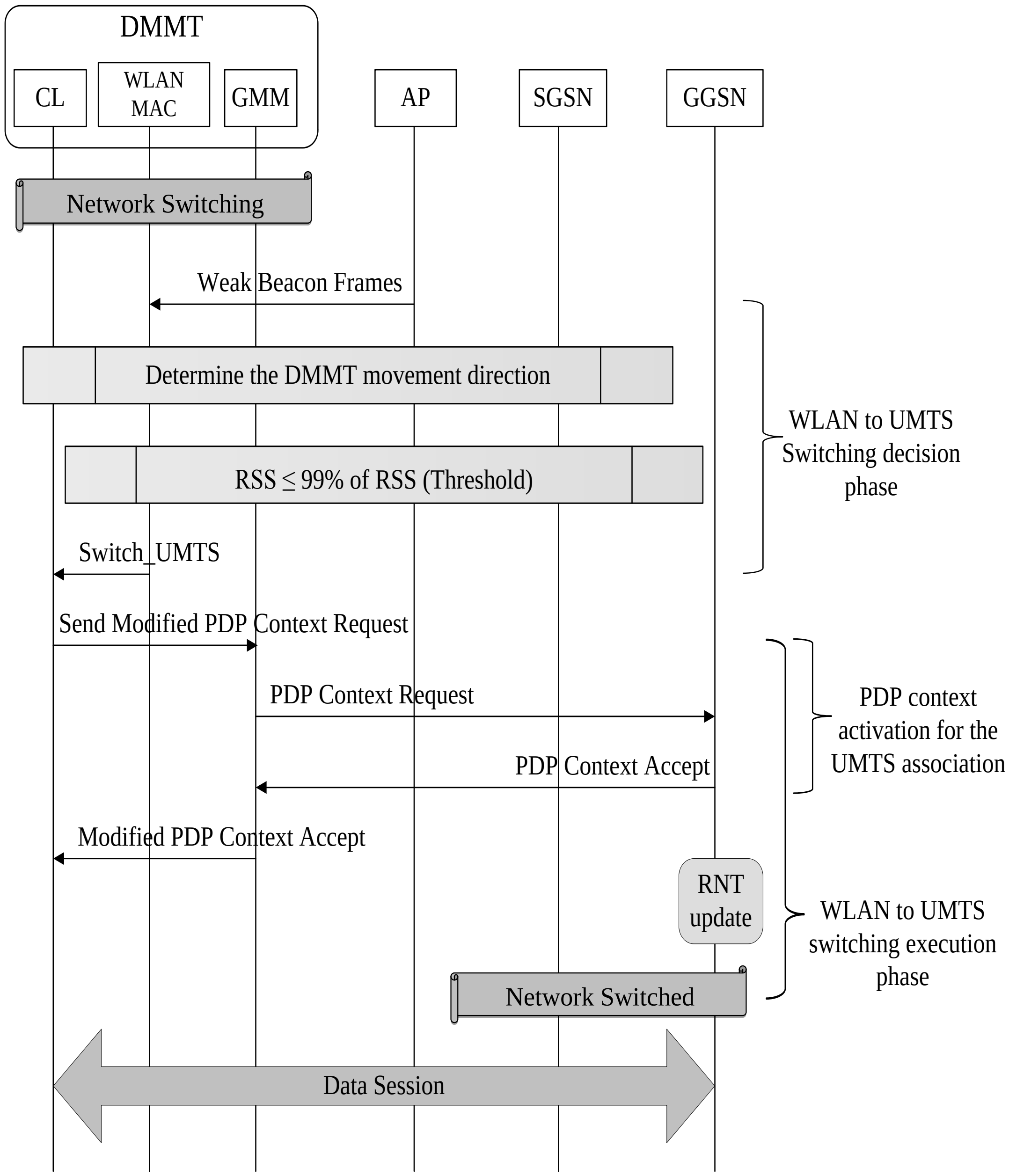

Figure 3.25: SVHOP for WLAN to UMTS switching

UMTS network by sending Modified PDP context request. As explained in Section 3.7.1, in order to initiate or continue a data session via UMTS network the DMMT has to establish a PDP context to the UMTS gateway, i.e., UMTS GGSN. To keep it simple, only the PDP context activation request and accept are shown in the Figure 3.25 and other messages like RAB assignment and RB setup is suppressed, although we implemented them in the simulations.

The GMM module which also has the functionalities of SM sends a Modified PDP context activation request message to the GGSN by inserting the UMTS_IP address in the PDP address field. Actually, at this stage this UMTS_IP is basically provided by the DMMT CL. This is the same UMTS_IP address which was acquired, when the DMMT established its connectivity to the UMTS network first time. On receipt, the GGSN will search and find a matching entry in its RNT, which is mapped with the WLAN_IP. This will be a clear indication that this is the same DMMT who initiated its session in the UMTS network, and then it moved to the WLAN network, now returning back to its home UMTS network. Therefore, it is desired that after performing the PDP context, the PDUs should be sent via Gn interface. Henceforth, after performing the PDP context activation, the PDUs will be sent to the DMMT from UTRAN.

3.8 Performance Analysis during VHO

This section presents the proposed SVHOP vertical handoff performance analysis for both upward and downward vertical handoff scenarios. For the comparison purposes, in addition with the SVHOP, the base line protocols i.e., MIPv6 and tight coupling network integration protocols are also evaluated.

3.8.1 Downward VHO

Downward VHOs (UMTS to WLAN) are basically opportunity based VHO. In this type of handoff MN performs VHO to the target network even when the current network is still available. Therefore, depending on the user preference, the user may or may not switch to the other network. Since, an overlaid network is always present; therefore, handoff timing is not often crucial in downward VHOs. Such VHO is performed to achieve high data rate and low cost services [38].

The Mobile IP handoff procedure is composed of layer 2 (L2) and layer 3 (L3) delays [18, 50, 54]. With L2 handoff, the MN changes its physical link layer connection with the current access point and continues its ongoing session with the new access point. After the successful L2 connection establishment with the new access point, a L3 handoff is carried out. For L3 handoff, the MN detects its network changing by extracting the new subnet prefix information from RAs. Therefore, a L3 handoff is always performed after establishing a link layer connection with the new access network. Therefore, mathematically the total MIPv6 handoff delay (D_{MIPv6}) can be represented as the sum of layer 2 and layer 3 handoff delays [18, 50, 54].

$$D_{MIPv6,\, UMTS\text{-}WLAN} = D_{L2} + D_{L3} + T_{data} \tag{3.6}$$

As illustrated in Figure 2.7 [4, 50], in the previous chapter, the layer 3 delay can further be broken down as:

$$\begin{aligned} D_{MIPv6,\, UMTS\text{-}WLAN} &= D_{L2} + D_{RD} + D_{CoA\text{-}Config} + D_{DAD} \\ &\quad + D_{HA\text{-}REG} + D_{RR} + D_{CN\text{-}REG} + T_{data} \end{aligned} \tag{3.7}$$

Where, D_{RD} represents the latency to perform the route discovery process. $D_{CoA\text{-}Config}$ is the time required by the MN to configure the globally unique IPv6 address with the help of address configuration rule such as EUI164. D_{DAD} shows the time required to ensure the uniqueness of the configured CoA on the foreign network. $D_{HA\text{-}REG}$ and $D_{CN\text{-}REG}$ illustrates the delay required to perform the CoA registration procedure with HA and CN, respectively. D_{RR} is the return routability delay.

Table 3.1: List of notations

Notations	Meanings
T_{RS}	Transmission delay required to send the RS message
T_{RA}	Transmission delay required to send the RA message
$T_{HA\text{-}BU}$	Transmission delay required to send the BU message to HA
$T_{HA\text{-}BA}$	Transmission delay required to send the BAack by HA
P_{HA}	Processing delay required by HA to process the binding updates
$T_{CN\text{-}BU}$	Transmission delay required to send the BU message to CN
$T_{CN\text{-}BA}$	Transmission delay required to send the BAack by CN
P_{CN}	Processing delay required by CN to process the binding updates
T_{HoTI}	Transmission delay required to send the HoTI message
T_{CoTI}	Transmission delay required to send the CoTI message
T_{HoT}	Transmission delay required to send the HoT message
T_{CoT}	Transmission delay required to send the CoT message
λ_P	Packet arrival rate (packets/second)
T_{data}	One way delay required to send first PDU from IS to DMMT at new network

Authors in [50] suggested that the DAD process can be avoided to reduce the MIPv6 handoff latency, if the address duplication probability is low because of the controlled networking environment. Therefore, by omitting the D_{DAD} delay component from the above expression around 1 second long latency can be avoided from the MIPv6 delay calculation. Table 3.1, represents the further break downed MIPv6 parameter's notations and meanings. In order to perform an in-depth evaluation of the transmission and processing delay involved by remaining MIPv6 components, the above expression can be written as:

$$D_{MIPv6,\,UMTS\text{-}WLAN} = D_{L2} + (T_{RS} + T_{RA}) + D_{CoA\text{-}Config} + (T_{HA\text{-}BU} + T_{HA\text{-}BA} + P_{HA}) + (T_{HoTI} + T_{CoTI} + T_{HoT} + T_{CoT}) + (T_{CN\text{-}BU} + T_{CN\text{-}BA} + P_{CN}) + T_{data} \quad (3.8)$$

The tight coupling vertical handoff mechanism is exclusively based on the link layer. This simply means that when the wireless device moves to the foreign access network no IP layer interaction is required. Let $T_{ME\text{-}RNCE}$ and $T_{RNCE\text{-}SGSN}$ be the transmission delay required to send the vertical handoff messages between ME to RNC emulator and RNC emulator to SGSN, respectively. The P_{RNCE} and P_{SGSN} represent the processing cost required by the RNC emulator and the SGSN to act in accordance with the received signaling message. These processing delays occurred because the SGSN has to establish the GTP tunnel with the RNC emulator to send and receive the PDUs while the wireless device is moved to the foreign network. Since, three messages are exchanged between the ME and the SGSN to perform the vertical handoff, namely, handoff initiation, response and complete. Therefore, the total HO delay equation can be expressed as:

$$D_{TC,\,UMTS\text{-}WLAN} = 3(D_{ME\text{-}SGSN}) + P_{SGSN} + P_{RNCE} + T_{data} \quad (3.9)$$

Since, the RNCE relays the handoff initiation, response and complete messages. Therefore, the $D_{ME\text{-}SGSN}$ will consist of the transmission delay between the ME and RNCE ($T_{ME\text{-}RNCE}$) and RNCE to the SGSN ($T_{RNCE\text{-}SGSN}$). Hence, equation 3.9 can now be expressed as:

$$D_{TC,\,UMTS\text{-}WLAN} = 3(T_{ME\text{-}RNCE} + T_{RNCE\text{-}SGSN}) + P_{SGSN} + P_{RNCE} + T_{data} \quad (3.10)$$

On the other hand, from the Figure 3.24, the total handover delay of the SVHOP can analytically be computed as:

$$D_{SVHOP,\,UMTS\text{-}WLAN} = D_{L2} + S_{Notification} + P_{GGSN} + T_{data} \quad (3.11)$$

Where, D_{L2} represents the WLAN link establishment delay. The $S_{Notification}$ is the signaling cost required to send the *WLAN Association Notification* to the GGSN

($T_{DMMT\text{-}GGSN}$) and receive back the acknowledgement ($T_{GGSN\text{-}DMMT}$). The P_{GGSN} shows the processing delay required by the GGSN to update its RNT with the double IP address entry provided by the DMMT so that all the subsequent PDUs can be sent to the roamed DMMT without any service interruption. Therefore, the above expression can be written as:

$$D_{SVHOP,\ UMTS\text{-}WLAN} = D_{L2} + 2T_{GGSN\text{-}DMMT} + P_{GGSN} + T_{data} \tag{3.12}$$

It is worth mentioning here that in the proposed SVHOP mechanism, we enabled the multi-homing functionalities in DMMT. Therefore, when the DMMT is located in an overlapping zone, meanwhile establishing link layer connection with the WLAN network by WLAN_MAC module and sending the *WLAN Association Notification* to the GGSN, the DMMT keeps its session alive from the UTRAN by its RLC_MAC module and continuously receiving the PDUs without any service interruption. Consequently, the handoff delay will depend on the time required by the GGSN to update its RNT and the signaling cost associated with the *WLAN Association Notification Reply*.

Therefore, by omitting the D_{L2} and dividing the $S_{Notification}$ cost into the half, the total handoff delay of the SVHOP mechanism can be expressed as:

$$D_{SVHOP} = T_{GGSN\text{-}DMMT} + P_{GGSN} + T_{data} \tag{3.13}$$

By comparing equation 3.8, 3.10 and 3.13 we can conclude that the vertical handoff delay during UMTS to WLAN network switching will be:

$$D_{MIPv6,\ UMTS\text{-}WLAN} >> D_{TC,\ UMTS\text{-}WLAN} >> D_{SVHOP,\ UMTS\text{-}WLAN} \tag{3.14}$$

Since, the packet loss during vertical handoff directly depends on the handoff delay. Therefore, the packet loss can be stated as:

$$PL_{MIPv6,\ UMTS\text{-}WLAN} >> PL_{TC,\ UMTS\text{-}WLAN} >> PL_{SVHOP,\ UMTS\text{-}WLAN} \tag{3.15}$$

3.8.2 Upward VHO

In upward VHO, UE is served with the high data rate and low cost network; it needs to perform VHO because it is moving away from the current network coverage area, e.g., from WLAN to UMTS network. Hence, in such type of VHO timing is crucial. An early handoff results in unnecessarily high cost and low data rate services of the target network, whereas a late VHO results in packet loss and degradation of service [38].

In case of MIPv6 WLAN to UMTS VHO computation, the DAD delay is not included. This is because, for the integrated UMTS/WLAN network, the UMTS and WLAN networks are considered as home and foreign networks, respectively. As mentioned in [50, 53], when the MN returns back home network, prior to its current binding expiration for its home address, the DAD procedure will not be performed. Moreover, in returning home case, sending the HoTI message exclusively is sufficient. Hence, no CoTI message is required in the return routeablity process [53]. Therefore, the expression of MIPv6 delay computation for WLAN to UMTS can be represented as:

$$\begin{aligned} D_{MIPv6,\,WLAN\text{-}UMTS} = {} & D_{L2} + (T_{RS} + T_{RA}) + D_{CoA\text{-}Config} + \\ & (T_{HA\text{-}BU} + T_{HA\text{-}BA} + P_{HA}) + (T_{HoTI} + T_{HoT}) \\ & + (T_{CN\text{-}BU} + T_{CN\text{-}BA} + P_{CN}) + T_{data} \end{aligned} \tag{3.16}$$

In case of WLAN to UMTS tight coupling based vertical handoff scenario, the delay computation would be quite similar as it was in case of UMTS to WLAN. Nevertheless, only few delay components will be replaced. For example, as the wireless device moves from the WLAN to the UMTS network, therefore, the GTP tunnel will be set up between the SGSN and RNC. If P_{RNC} be the processing delay required by the RNC to maintain the GTP tunnel then the total vertical handoff delay required to complete the VHO in case of tight coupling mechanism can be expressed as:

$$\begin{aligned} D_{TC,\,WLAN\text{-}UMTS} = {} & 3(D_{ME\text{-}RNCE} + D_{RNCE\text{-}SGSN}) + P_{SGSN} + \\ & P_{RNC} + T_{data} \end{aligned} \tag{3.17}$$

From Figure 3.25, the VHO delay of the SVHOP mechanism can be express as:

$$D_{SVHOP,\ WLAN\text{-}UMTS} = D_{PDP} + P_{GGSN} + T_{data} \tag{3.18}$$

Where, D_{PDP} is the delay required to perform the PDP context activation process. The P_{GGSN} is the processing delay required by the GGSN to update the RNT table. However, unlike the UMTS to WLAN handover case, this time GGSN will update its RNT by deleting the matching IP addressing entry in its table.

In contrast to the MIPv6 and tight coupling mechanism, in the proposed SVHOP the proactive RSS algorithm is implemented. Therefore, before reaching to the WLAN actual boundary the SVHOP triggers the VHO at the virtual boundary of WLAN. Moreover, since the multi-homing feature is implemented, therefore, the SVHOP mechanism utilizes its both physical interfaces during handoff. Therefore, even the DMMT is receiving PDUs from WLAN access network, the DMMT UMTS interface starts establishing its connection with the UTRAN. Due to the parallel transmission, the VHO delay will not be influenced by the PDP context activation. Hence, only the processing delay required by the GGSN for switching the interfaces and the one way delay required to send the PDU at the new network influences the handoff delay. Therefore, the total handover delay for the SVHOP mechanism can analytically be computed as:

$$D_{SVHOP,\ WLAN\text{-}UMTS} = P_{GGSN} + T_{data} \tag{3.19}$$

By comparing equation 3.16, 3.17 and 3.19 we can conclude that the vertical handoff delay during WLAN to UMTS network switching will be:

$$D_{MIPv6,\ WLAN\text{-}UMTS} >> D_{TC,\ WLAN\text{-}UMTS} >> D_{SVHOP,\ WLAN\text{-}UMTS} \tag{3.20}$$

Since, the packet loss during vertical handoff directly depends on the handoff delay. Therefore, the packet loss can be stated as:

$$PL_{MIPv6,\ WLAN\text{-}UMTS} >> PL_{TC,\ WLAN\text{-}UMTS} >> PL_{SVHOP,\ WLAN\text{-}UMTS} \tag{3.21}$$

3.9 Chapter Summary

In this chapter, the proposed SVHOP mechanism, objectives, design considerations and evaluation are discussed in detail. Several problems that were not properly addressed in the existing mobility management protocols are also highlighted. Moreover, how these issues are addressed by implementing the suggested techniques in the wireless client device and UMTS GGSN are presented. This chapter shows that the proposed SVHOP mechanism is an optimal choice to integrate the UMTS and WLAN networks.

The seamless mobility is attained by optimizing the signaling and process cost during vertical handoffs. The multi-homing technique along with the parallel signaling transmission mechanism and proactive RSS based mechanism are applied to minimize the vertical handoff latency and packet loss. To understand the significance of the proposed technique in terms of seamless mobility management, a comprehensive performance analysis of SVHOP during VHO is presented along with the existing mobility management protocols. It has been observed that the proposed mechanism is an optimal choice to attain the seamless mobility.

In addition to the seamless mobility, the ease of network implementation is a very important factor to be taken into consideration for any sophisticated integration mechanism. Having identified this, the proposed mechanism is designed in a manner that the proposed network architecture does ensure that significant modification to the existing UMTS and WLAN networks can be avoided. More precisely, only the UMTS GGSN needed to be upgraded with the suggested CL module.

For an APO free mechanism, instead of adding the overhead information into the PDUs which eventually decreases the network throughput and increases the application response time, the existing IP address fields are reused by implementing IP swapping mechanism. In addition, it is demonstrated that the proposed IP address swapping mechanism also helps in achieving the transparency of the upper layer and correspondent node.

CHAPTER 4

RESULTS AND DISCUSSION

4.1 Chapter Overview

This chapter presents the analysis and evaluation of the proposed SVHOP protocol based on the proposed methodology and designed considerations elaborated in the previous chapter. Different simulation scenarios are designed using OPNET Modeler tool and the network performance is analyzed by implementing both real and non-real time data applications. The performance metrics along with the simulation parameters are also discussed in this chapter. In all the simulation scenarios, the performance of the proposed SVHOP protocol is compared with the MIPv6 and tight coupling UMTS/WLAN integration protocols. This chapter also includes the qualitative analysis of the proposed, baseline and several other leading integration protocols.

4.2 Simulation Network Design

This section illustrates the simulation network topology for the integration of UMTS and WLAN networks. The network design consists of three major parts: UMTS network, WLAN network and Internet Service Provider (ISP), as shown in Figure 4.1. It should be noted that in the network design only UMTS PS domain is considered and UMTS CS domain is neglected for simplicity. The UMTS network is composed of Node-B, Radio Network Controller (RNC), SGSN and GGSN. The RNC is connected with the Node-B and SGSN with the ATM OC-3 link that supports data rate up to 155.52 Mbps. The GGSN is connected to the SGSN with the PPP DS-3 bi-directional link that supports data up to 44.736 Mbps.

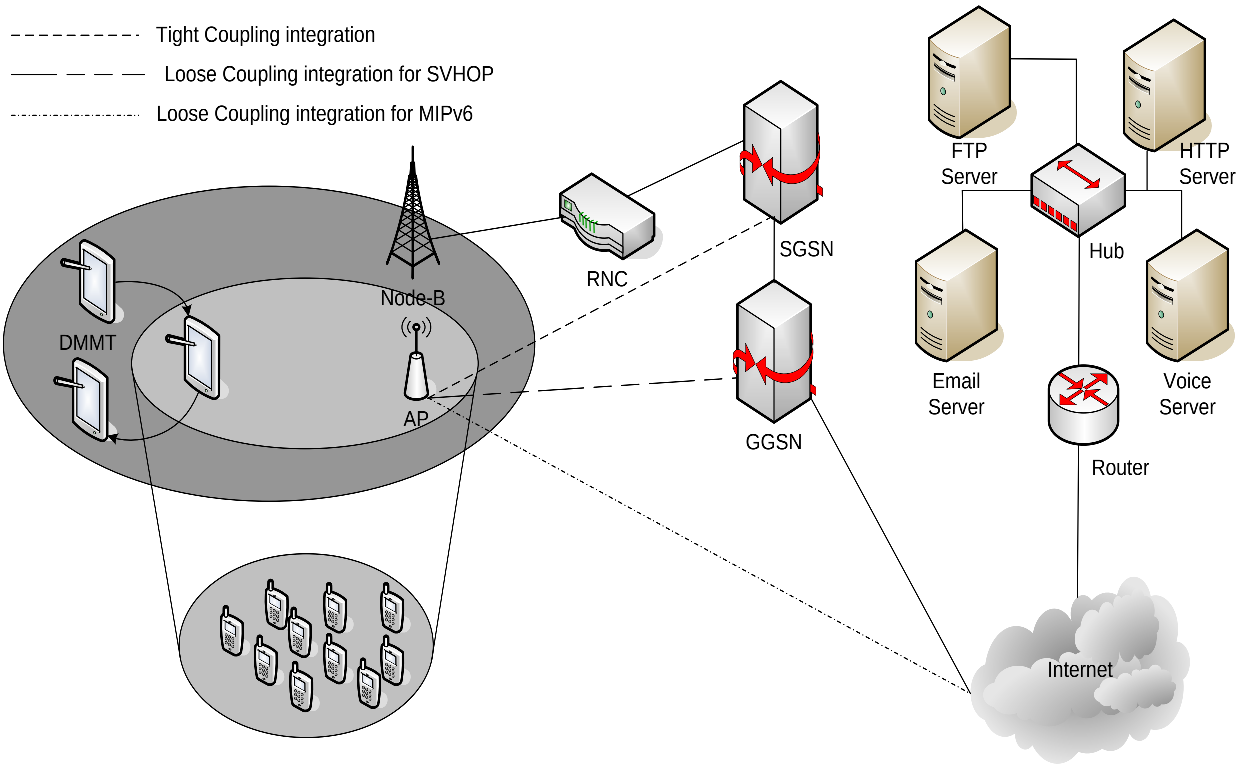

Figure 4.1: Simulation network design

For the evaluation and comparison purposes of SVHOP, MIPv6 and tight coupling mechanism, the WLAN AP is connected to the UMTS network via different points. For the SVHOP, as illustrated in Figure 4.1, the WLAN AP is connected with the GGSN. For the tight coupling, the WLAN is connected with the UMTS SGSN. In contrast with the SVHOP and tight coupling mechanisms in which the WLAN AP is connected with the UMTS core network, the MIPv6 connects the WLAN AP with the internet cloud to establish an indirect connection with the UMTS network. The WLAN AP is located within the UMTS coverage area. This simulation scenario reflects a real-world scenario where WLAN is operated as a hotspot under the coverage of UMTS cell. Such hotspots serve airports, campuses, buildings, train stations, hotels, etc. In addition, the Voice over Internet Protocol (VoIP), FTP, Email and HTTP servers are located at the back of the internet cloud to provide data services to the DMMT.

4.2.1 Simulation Scenarios

Two different simulation scenarios have been designed and evaluated separately for the evaluation of the SVHOP and benchmark protocols. The main difference of these scenarios is either the DMMT is moving back and forth between the UMTS and WLAN access networks or it is statically located inside the integrated WLAN access network. For the mobile DMMT, handoff latency, session blackout time, transient packet loss and lost information during the handoff are measured for upward and downward VHO cases. On the other hand, when the static DMMTs are located inside the WLAN access network the additional protocol overheads, application response time, and system throughput has been measured.

4.2.1.1 Mobile DMMT

It is assumed that the DMMT always initiates its data session via UMTS access network. For the DMMT movement, a ping-pong trajectory that produces ten back and forth motion across the UMTS and WLAN network is defined. For simplicity, as illustrated in Figure 4.2, let us consider a single UMTS to WLAN and WLAN to

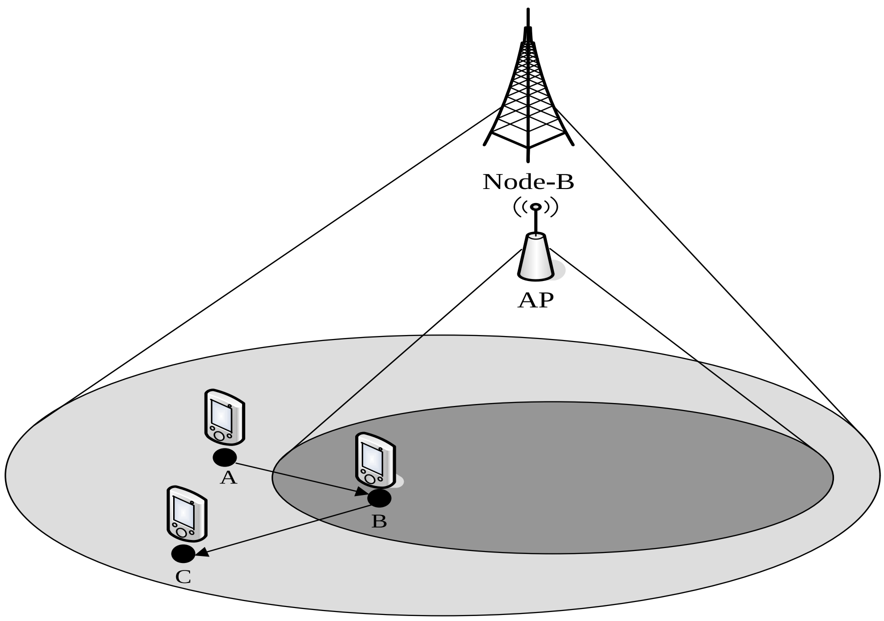

Figure 4.2: The back and forth movement of DMMT between UMTS and WLAN access networks

UMTS handoff scenario. In this case, the trajectory will contain three points, say A, B and C. The point A locates the initial point of the simulation, from where the wireless device activates its data session by requesting the GMM attach and PDP context activation with the UMTS network. After the successful establishment of the UMTS data session, the wireless client moves towards point B (which is located inside the overlaid WLAN access network). When the wireless device receives the initial beacon messages it will perform the downward vertical handoff according to the condition specified in Section 3.5.3. After reaching to the point B, the wireless device again starts moving back to the UMTS access network. After crossing the edge of the WLAN access network the wireless client performs the upward vertical handoff. It must be noted that when comparing the proposed mechanism, MIPv6 mechanism and the tight coupling mechanism, separate simulation scenarios are designed and evaluated in which the upward and downward vertical handoff follow the particular vertical handoff protocol.

4.2.1.2 Stationary DMMT

For the stationary DMMTs scenario, it is assumed that the DMMT initiates its data inside the overlaid WLAN coverage region of the integrated UMTS/WLAN network. Because of no movement, all the PDUs are served by the WLAN access network. This simulation scenario represents how efficiently the WLAN is operated when it is coupled with the UMTS network by using SVHOP and baseline protocols. In terms of performance metric, this simulation scenario illustrates the effectiveness of the SVHOP and benchmark protocols by analyzing the additional protocol overheads, application response time, and system throughput when the wireless client is located in the foreign network.

4.2.2 Simulation Parameters

The simulation parameters for the MIPv6, tight coupling and SVHOP are illustrated in Table 4.1. The integrated UMTS/WLAN network along with the ISP servers is

Table 4.1: Simulation parameters

Parameters	**Values**
Simulation area	1000 m * 1000 m
Simulation duration	Variable
Access Point Transmitter Power	20dBm (100mW)
DMMT WLAN interface threshold	-95dBm
UMTS data rate	2 Mbps
WLAN data rate	11 Mbps
DMMT velocity	0 or 1.5 m/s
Path loss index (η)	3
Constant power loss in wireless channel (L)	55 dB
AP Frequency	2.4 GHZ
WLAN actual boundary	100m
WLAN virtual boundary	95m
Link between RNC-Node B	ATM OC-3 (155.52 Mbps)
Link between RNC-SGSN	ATM OC-3 (155.52 Mbps)
Link between GGSN-SGSN	PPP DS-3 (44.736 Mbps)
Link between Servers and Hub	100Base-T (100Mbps)

deployed in the area of 1000 * 1000 meters. It is assumed that the path loss of the wireless channel decays according to the LNSM in which the constant power loss in the propagation medium is 55 dB [89] and path loss index is 3 [86]. The WLAN AP default power transmission is set as 100mW (20dBm) [87]. Moreover, the WLAN interface receiver sensitivity threshold is -95dBm. The IEEE 80.11b operates in license-exempt band i.e., 2.4 GHZ and provides the data rate up to 11Mbps [31]. On the other hand, UMTS supports 2Mbps for mobile users. For the mobile DMMT, the pedestrian trajectory speed is implemented which is 5.0 kilometers per hour (km/h) or approximately 1.5 meters per second (m/s).

4.2.3 Simulation Traffic

In the simulation scenarios, the DMMT communicate with the internet FTP, HTTP, voice and E-mail servers. Table 4.2 represents applications and measurement parameters tested for the integrated UMTS-WLAN networks. These applications match different UMTS QoS classes. A conversational class represents real time traffic flows such as VoIP. The background class represents both FTP and E-mail services and interactive class corresponds to web browsing (HTTP). The streaming class represented by the video streaming.

Table 4.2: Description of the tested applications and measurement parameters

Application	**QoS Class**	**Measurement Parameter**	**Protocol**
VoIP services	Conversational	Handoff delay, session blackout time, packet loss, lost information, additional protocol overheads	UDP
E-mail	Background	Download response time and upload response time	TCP
FTP	Background	Download response time, upload response time and system throughput	TCP
HTTP	Interactive	Page response time	TCP
Video	Streaming	End to end delay	UDP

As illustrated in Table 4.3, in order to evaluate the seamless mobility management performance, five different codec types have been implemented in the integrated UMTS/WLAN network. Namely, these are G.711, GSM, G.723.1, G.726 and G.729.

Table 4.3: Codec types and their associated parameters

Codec type	**Data rate (kbps)**	**Payload size (bytes)**	**Packets per second**
PCM (G.711)	64	80	100
GSM	13.2	33	50
G.726	32	40	100
G.729	8	10	100
G.723.1	5.3	20	33.3

The G.711, also known as Pulse Code Modulation (PCM), operates at the data rate of 64 kbps [90]. It attains the high Mean Opinion Score (MOS) of 4.2. The MOS ranges from 1 (bad quality voice) to 5 (excellent quality voice). Generally, PCM is considered a base standard of codec. Moreover, currently, the G.711 codec is the most widely supported and implemented codec in IP telephony because of its ability to provide high voice quality in IP networks. The G.723.1 which operates at the data rate of 5.3 kbps [91], whereas, the GSM FR operates at the data rate of 13.2 kbps. The packets per second of GSM and G.723.1 are 50 and 33.3, respectively. The data rate of G.729 is 8kbps [92]. The G.729 sends 100 packets per second; each packet contains the payload size of 10 bytes. The G.723.1, G.729 and GSM are the most widely used codecs in GPRS and UMTS networks. The G.726 sends data at 32kbps with the payload size of 40 bytes each at 100 packets/sec.

4.2.4 Simulation Assumptions

The following assumptions have been made for the assessment of integrated UMTS/WLAN network.

- In order to predict the path loss behavior of radio waves during propagation, Log Normal Shadowing Model has been implemented.
- The mobile DMMT follows the predefine ping pong trajectory as defined in Section 4.2.1.1
- No packet buffering mechanism is implemented.
- For the VoIP services, no silence suppressed mechanism is applied in the simulation scenarios.
- No header compression mechanism is used.

4.2.5 Simulation Metrics

When performing the quantitative analysis of integration protocols, this research concentrates on several metrics which include handoff delay, session blackout time, transient packet loss during handoffs, lost information, additional protocol overheads, application response time, system traffic received (throughput). These matrices are used in this research effort to evaluate the performance of proposed SVHOP with the other existing leading integration protocols.

4.2.5.1 Handoff Delay

The handoff latency is the most prominent parameter in analyzing the performance of an integration protocol. In case of integrated UMTS/WLAN network, the handoff latency of a roaming MN can be measured as the difference of the time when the DMMT receive the first PDU from the new access network (T_{NN}) to the time when the DMMT received the last PDU from previous network (T_{PN}). Mathematically, the VHO delay produced by any specific protocol (D_{VHO}, x) can be represented as:

$$D_{VHO,x} = T_{NN} - T_{PN} \tag{4.1}$$

It is worth recalling here that when the MN disconnects its session with the previous network and wants to connect its ongoing session with the new network, there are several vertical handoff signals required to be sent among the network nodes. In addition, all the network nodes participating in the handoff procedure produces some additional delay to perform in accordance with the received message. Therefore, the signaling and processing cost highly influences the vertical handoff delay computation. As discussed in Section 3.2.1.1, if more signals are exchanged and more nodes are participated to perform a handoff then the HO delay will be higher. Similarly, if less signaling is performed and less number of nodes contributes in the handoff process, a faster handoff can be executed. A detailed discussion of all individual delay components that influence the vertical handoff computation were discussed in detail in Section 3.7.

4.2.5.2 Session Blackout Time

During the handoff period the wireless clients are not connected with any of the integrated access network, therefore, all the data during this period is lost. In most of the cases, while roaming across different integrated overlaid heterogeneous wireless access networks, the wireless client performs several handoffs during the course of a single data session. Therefore, it is very important to measure the total session down time. The metric that illustrates the total amount of time in which no data can be sent or receive by the wireless clients for a single active data session due to the multiple handoffs, is termed as the Session Blackout Time (*SBT*). The session blackout time directly depends upon the product of handoff latency and number of times the wireless client performs the handoff per session. In an ideal case, if no handoff is performed during a data session, there should be no session blackout. Mathematically, the session blackout time can be expressed as:

$$SBT \ = D_{VHO,x} * \left(\frac{Number\ of\ handoffs}{session}\right) \tag{4.2}$$

4.2.5.3 Packet Loss

The packet loss is measured as the sum of all the lost and dropped data packets during the data session blackout time. During the handoff period the intermediate nodes/data senders are not aware of the current location of the mobile node. Therefore, during this time all the data packets will be lost because either the network would have been routing the data packets to the previous network (since, the mobile node current location is not updated yet, consequently, such packets are lost due of the absence of targeted mobile node in the previous network) or the network is busy in updating the current location of the mobile node. It should also be noted that, in addition to the session blackout time, the packet loss directly depend upon the packet arrival rate (λp). Therefore, the packet loss will be high and low with the fast and slow packet arrival rate, respectively. Mathematically, the packet loss during SBT by any particular integration protocol can be express as:

$$PL_x = SBT * \lambda_p \tag{4.3}$$

4.2.5.4 Lost Information

Any sophisticated vertical handoff protocol must ensure the minimum data lost during the vertical handoff. The lost information is another very important metric and it must be taken into prime consideration while defining the integration protocol. Actually, the lost information metric evaluates the integration protocol one step ahead of the packet loss. In most of the cases, only the number of packet loss calculation alone is not sufficient to analyze the true efficiency of the integrated network. For example, consider that two different traffic streams are present in the network, say A and B. The Stream A (with high payload size in the data packet) and B (with low payload size in the data packet) having similar packet arrival rate and suffers the same amount of vertical handoff delay. In such cases, the packet loss will produce identical results. However, it is quite obvious that the amount of information in case of traffic stream A will be higher compared to stream B.

Consequently, in order to analyze the information loss in terms of actual lost data bytes/bits according to the particular traffic scheme, lost information metric is required. The Transient Lost Information (TLI) metric of any particular integration protocol can be expressed mathematically as:

$$TLI = D_{VHO,x} * \lambda_p * \text{Ɣs} \tag{4.4}$$

Where,

Ɣs represents the payload size of PDU

4.2.5.5 Additional Protocol Overhead

Additional protocol overhead is one of the prime metric to enhance the overall performance of data services and network throughput. Protocol overheads are the additional bytes required to append in the PDUs for the data routing. Bandwidth is a scarce resource, especially in a wireless environment, therefore, the process of overhead append is a big obstacle to attain the high data service performance. By introducing the overhead information in the data packets, not only the data transmission but the overall network throughput is highly influenced. In order to provide the optimal data performance, it is desired that any sophisticated integration protocol must avoid inserting its own overhead information in the data traffic. The additional protocol overhead for the protocol x (*APOx*) can be defined as the sum of all additional overhead bytes required to send/received the PDUs. Mathematically, it can be expressed as:

$$\mathrm{APO}_x = \sum_{k=1}^{n} \Gamma_{Additinal\_bytes} \tag{4.5}$$

Where,

Γ represents the additional overhead bytes in each PDU and

n represents the number of PDUs sent/received.

4.2.5.6 Application Response Time

The Application Response Time (*ART*) is a key metric to determine the user satisfaction with the overall network performance. Generally, in data communication the ART is defined as the time elapsed between the "put" requests send by the end user to the IS and in response receives the complete requested file from the IS. The connection setup signaling delay is also included in this time. Mathematically, it can be represented as:

$$ART = Connection\_setup\_time + (N * NRT) \tag{4.6}$$

Where,

N shows the number of packets transmitted from/to the internet server and

NRT represents network response time

The network response time is influenced by several factors that include the payload size, overall network bandwidth available to the user/application, and the packet end to end delay. It should be noted that the end to end delay is composed of the time required by the data packet to traverse from source to destination along with the sum of all intermediate node processing, queuing and transmission delays.

4.2.5.7 Network Throughput

In data networks, the system/network throughput is the average rate of successful payload delivery to all the network nodes via communication channel. However, it must be noted that maximum achievable throughput is significantly lesser compared to the actual connection rate. Because, the throughput is what the network nodes see after the overheads. The OPNET defines the global throughput (system throughput) as the average bytes per second forwarded to all the network nodes by their transport layer. The throughput is usually measured in bits per second or bytes per second. Mathematically, the system throughput can be expressed as:

$$System\_throughput = \frac{\sum_{i=1}^{n}(traffic\_received)}{second} \tag{4.7}$$

Where,

n represents the number of nodes located inside the network

4.3 Simulation Results

In this section the proposed SVHOP and the benchmark protocols performance has been evaluated in terms of handoff delay, session blackout time, transient packet loss, lost information during handoff, additional protocol overheads, application response time and system throughput.

4.3.1 Mobile DMMT Scenario

Figure 4.3 and Figure 4.4 illustrates the downward and upward vertical handoff delay, respectively, when MIPv6 has been implemented on the integrated UMTS/WLAN network. In order to evaluate the vertical handoff delay, the graph plots the simulation run time with the corresponding DMMT packet delay variation. It is quite obvious that because of the complex infrastructure design, high number of intermediate nodes and low bandwidth the packet delay variation of UMTS network is higher compared to the WLAN network. It can be observed from Figure 4.3 that after the handoff the wireless client receives the first PDU from the target network at 31.7 seconds, whereas, the last packet received from the previous network was at 30.1 seconds. Therefore the downward vertical handoff time is 1.6 seconds. On the other hand, the 4.4 illustrates that the upward vertical handoff delay is 2 seconds, since the first PDU from UMTS network is received at 124.6 seconds and the last packet from the WLAN was received at 122.6 seconds.

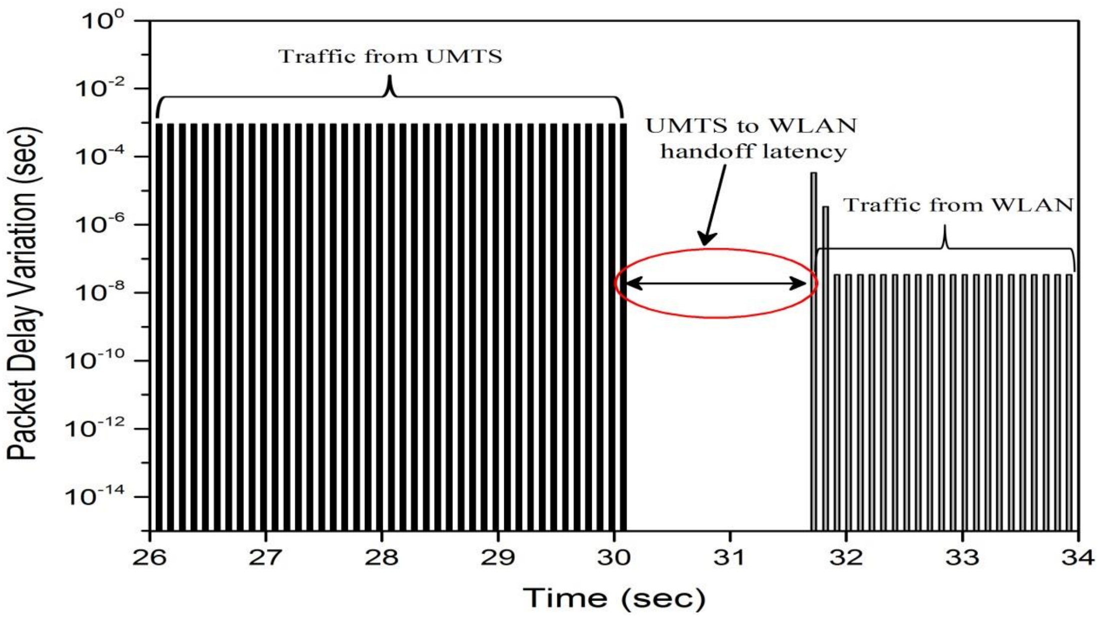

Figure 4.3: MIPv6: UMTS to WLAN handoff latency

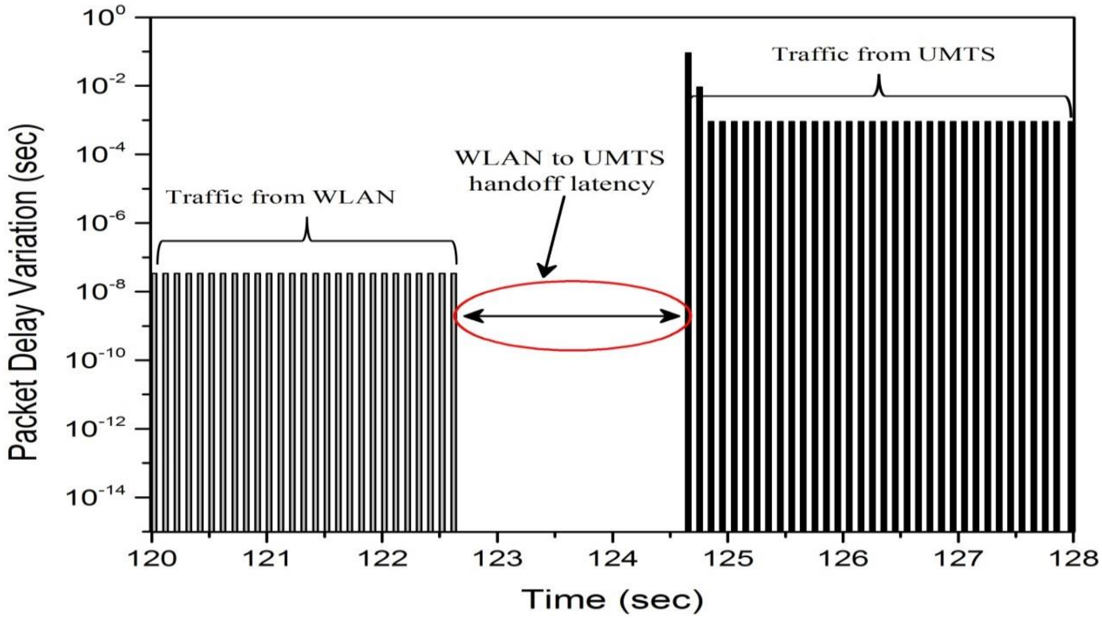

Figure 4.4: MIPv6: WLAN to UMTS handoff latency

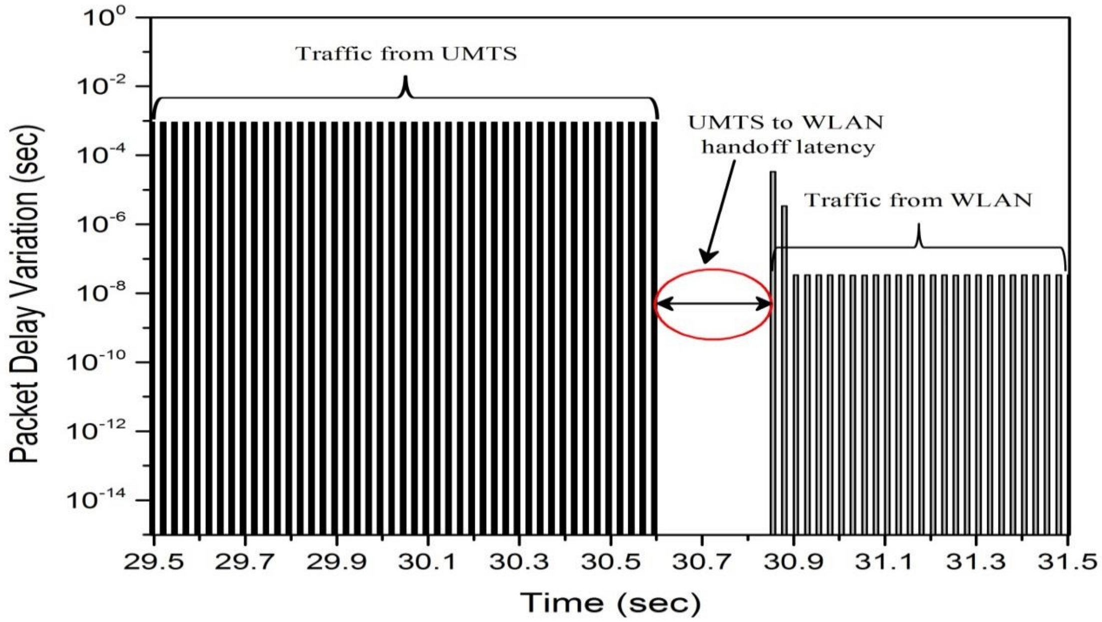

Figure 4.5: Tight coupling: UMTS to WLAN handoff latency

In case of tight coupling network integration mechanism, Figure 4.5 and Figure 4.6 shows the downward and upward vertical handoff delay, respectively. As dictated by the literature, tight coupling performs much faster vertical handoff compared to the network layer based vertical handoff solution.

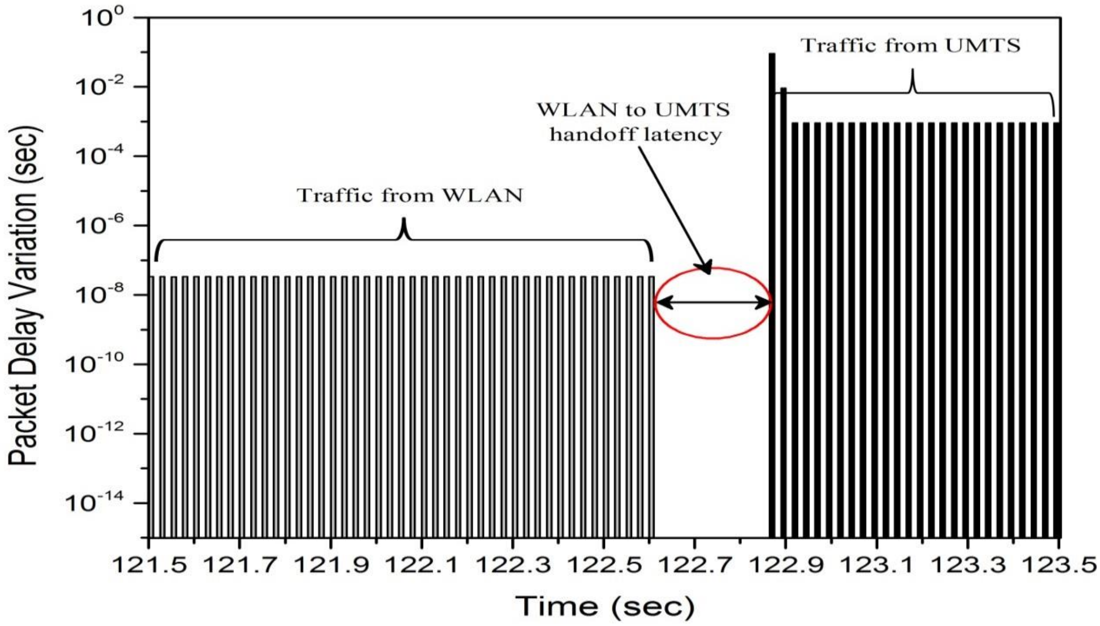

Figure 4.6: Tight coupling: WLAN to UMTS handoff latency

It can be observed from Figure 4.5 that after the handoff the wireless client receives the first PDU from the target network at 30.85 seconds, whereas, the last packet received from the previous network was at 30.6 seconds. Therefore, in tight coupling mechanism, the downward vertical handoff time is 250ms. On the other hand, the 4.6 illustrates that the upward vertical handoff delay is 275ms, since the first PDU from UMTS network is received at 122.875 seconds and the last packet from the WLAN was received at 122.6 seconds.

Figure 4.7 and Figure 4.8 demonstrates downward and upward vertical handoff delay in case of the proposed SVHOP mechanism, respectively. The VHO delay in case of UMTS to WLAN and WLAN to UMTS network is 125ms and 100ms, respectively. The analysis of MIPv6, SVHOP and tight coupling validates Equations 3.14 and 3.20 discussed in previous chapters that stated that the MIPv6 takes longest and SVHOP takes shortest time to execute VHO, whereas, tight coupling VHO delay lies in between these two protocols. Moreover, it should also be noted that the SVHOP meets the ITU G.114 recommendation, which suggested that that the one way end-to-end delay for the real time services should not exceed 150ms.

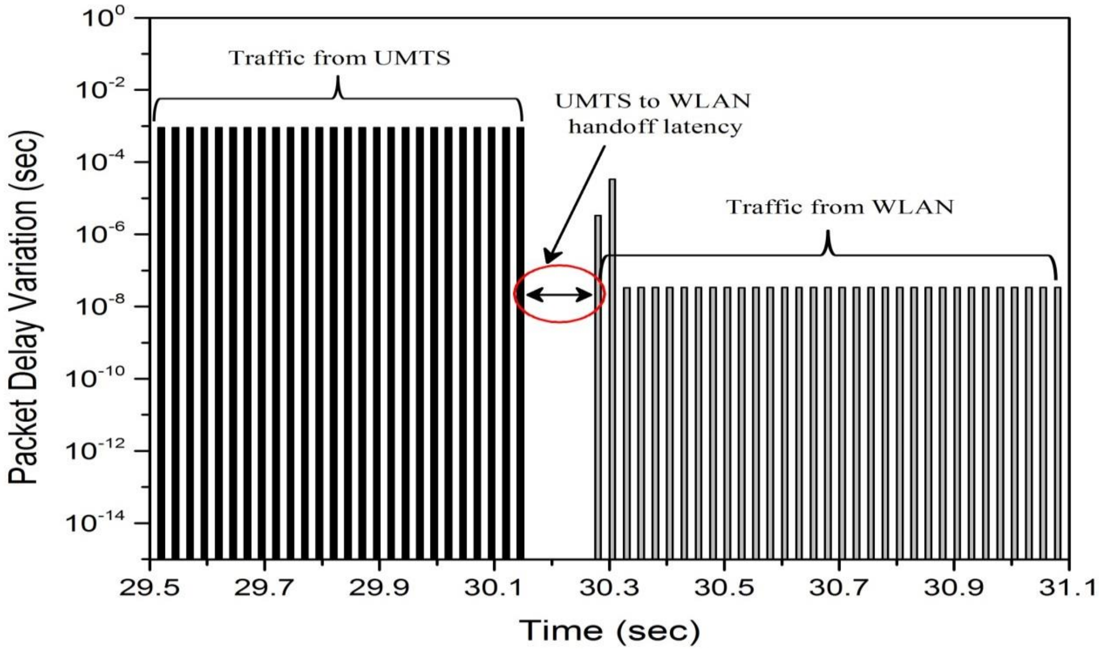

Figure 4.7: SVHOP: UMTS to WLAN handoff latency

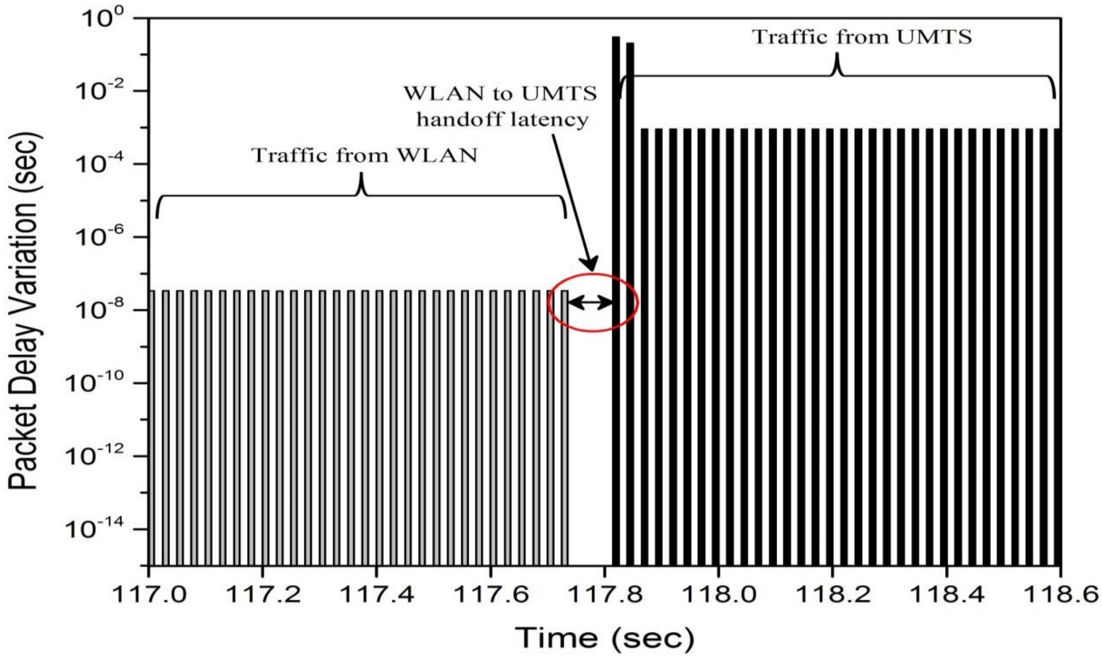

Figure 4.8: SVHOP: WLAN to UMTS handoff latency

Figures 4.9 and Figure 4.10 plot the session blackout time with the corresponding number of handoffs per session, while the DMMT is moving from the UMTS to WLAN and WLAN to UMTS network, respectively. All three, i.e., the proposed

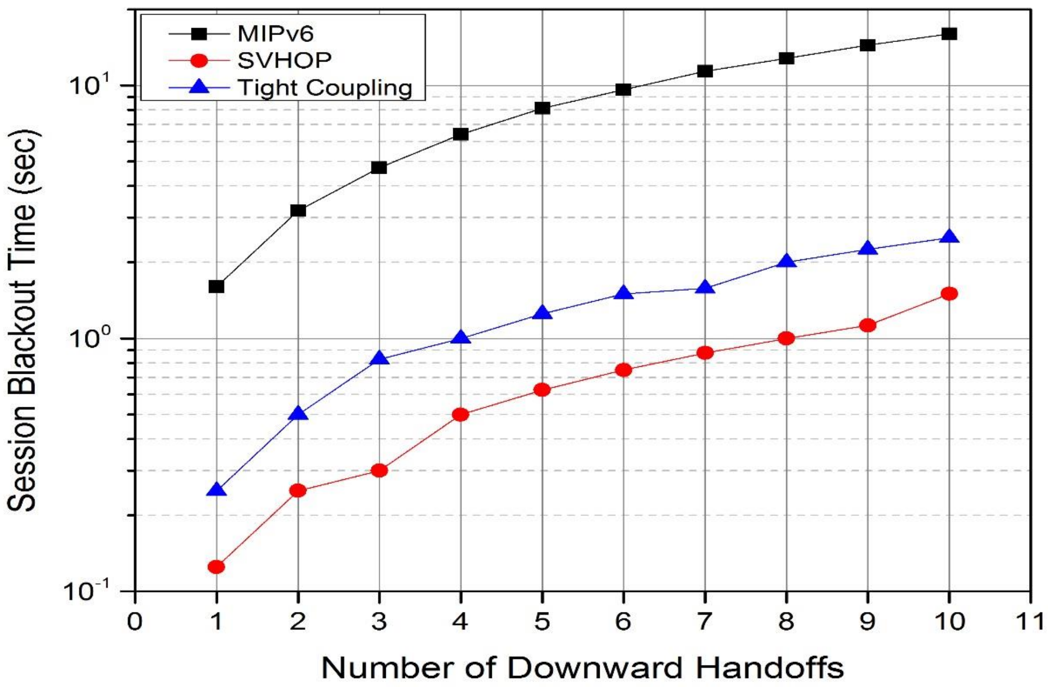

Figure 4.9: Session blackout time while the user is moving from UMTS to WLAN

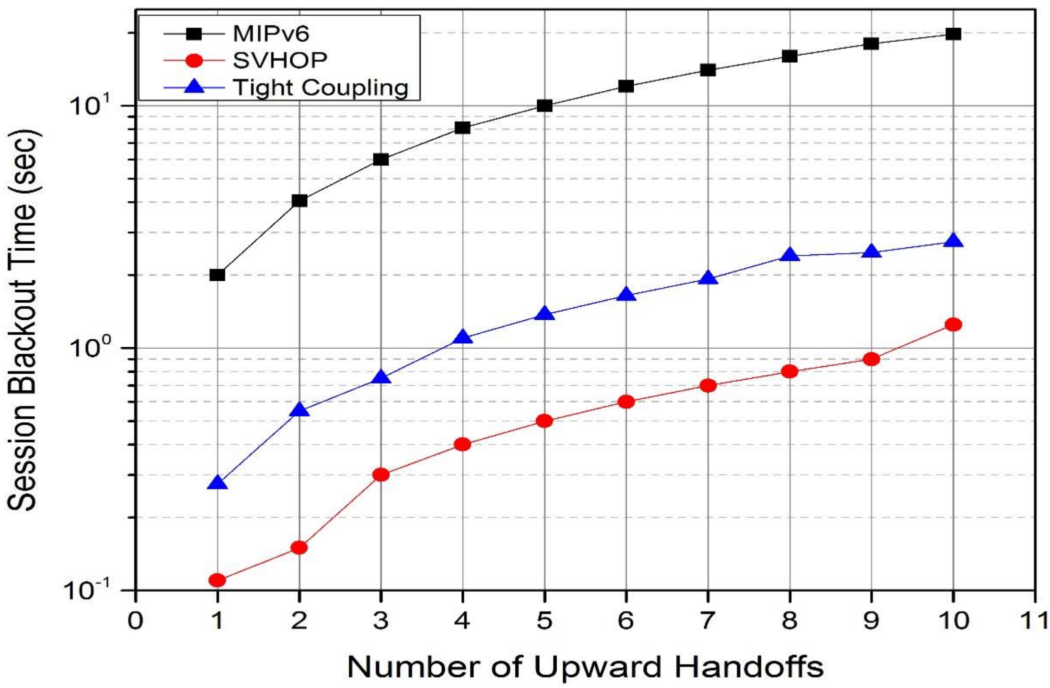

Figure 4.10: Session blackout time while the user is moving from WLAN to UMTS

SVHOP, tight coupling and MIPv6 protocols are analyzed. From Equation 4.2, it is clear that the total session blackout time directly depends upon the number of handoff performed by the wireless client during a session. Therefore, as the number of handoff per session increases, the total blackout time gradually increases. In general, we can observe that with the increase of the number of handoff, the MIPv6 influenced the most; conversely, the SVHOP affected the least.

Figures 4.9 and Figure 4.10, highlights some very interesting points. In case of MIPv6 and tight coupling mechanism, it should be noted that the blackout time for upward VHO is higher compared to the downward VHO case. However, for the proposed SVHOP the result is reversed. For MIPv6, the higher upward session blackout time occurred because of two reasons. First, the UMTS L2 handoff delay is much higher compared to the L2 delay of WLAN. Second, when the mobile terminal performs VHO from WLAN to UMTS network, the binding update and authentication with CN takes relatively higher transmission cost compared to the UMTS to WLAN case. This higher transmission cost is basically contributed by the higher number of UMTS network hops. For tight coupling, the higher upward session blackout time mainly occurred because of higher transmission cost of the first PDU when it is sent after the handoff from the UMTS network compared to the WLAN network. Consequently, similar to the MIPv6 case, the higher UMTS transmission cost increases the upward blackout time.

In contrast to the benchmarking protocols, in case of SVHOP mechanism the downward blackout time is higher compared to upward blackout time. This happened because of the implementation of the proactive RSS based algorithm in upward case. The proactive RSS protocol has been discussed in detail in Section 3.5.3. By using the proposed proactive RSS mechanism, even before reaching to the WLAN actual boundary the UMTS connection establishment mechanism was triggered by sending the PDP context activation request in parallel with the ongoing data session via the WLAN network. Consequently, as explained by using Equation 3.19 in the previous chapter, along with the first PDU transmission cost only the processing cost required by the GGSN to update its RNT table will influence the upward session blackout time. On the other hand, for the download blackout time as explained in Equation 3.13 in the previous chapter, in addition with the P_{GGSN}, half of the $S_{Notification}$ signaling cost also incorporated in the computation of total blackout time. Therefore, in SVHOP mechanism, compared to the downward case, an upward blackout time produces smaller data session breaks.

Figures 4.11 and Figure 4.12 plots the number of transient packet loss with different codec types used on y-axis, whereas, the x-axis contains the number of handoffs during a data session. Both UMTS to WLAN and WLAN to UMTS scenarios are represented in the above graphs. The collected statistics highlights two very interesting points. First, irrespective of the implemented protocol or the upward/downward simulation scenario, different codec produces varying transient packet losses. From Equation 4.3, we know that the packet loss directly depends upon the packet arrival rate/ packets per second. Table 4.3 demonstrated that the PCM, GSM and G.723.1 send 100, 50 and 33.3 packets per second, respectively. Therefore, when the packets from internet server travel towards the wireless client with different packet speed, it is quite obvious that the PCM and G.723.1 produces highest and lowest number of packet loss, respectively, whereas, GSM values lies in between the other two voice encoder mechanisms. The main objective of this varying codec type

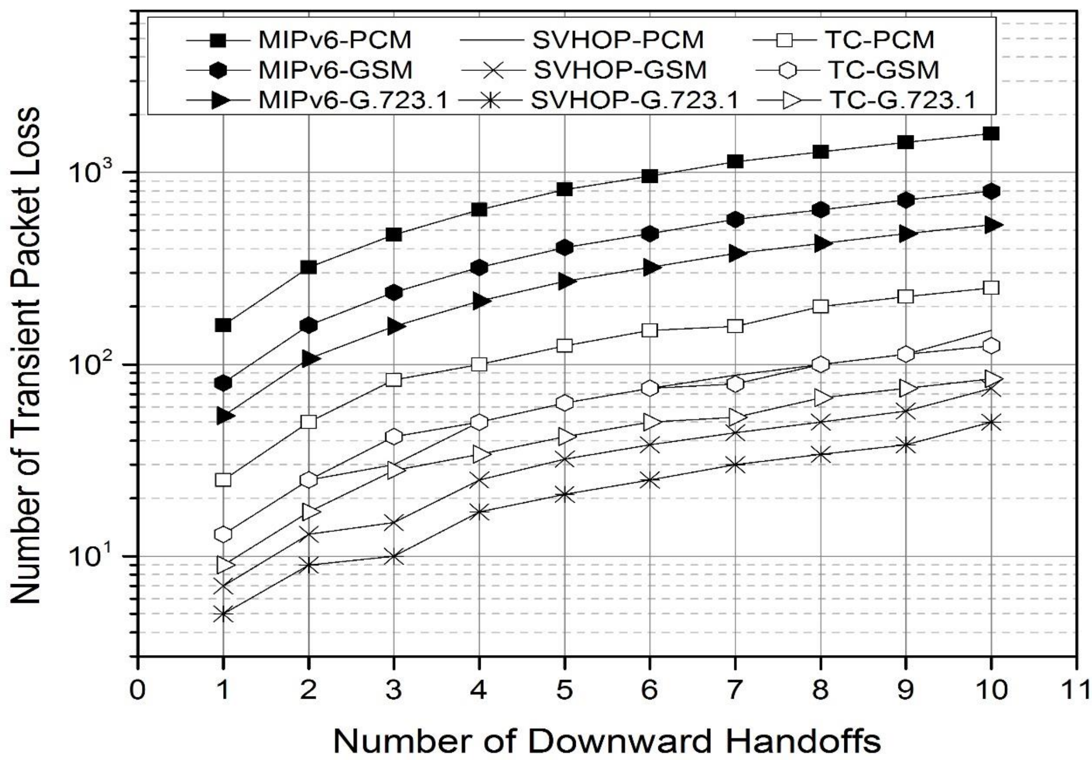

Figure 4.11: Number of transient packet loss during UMTS to WLAN handoff

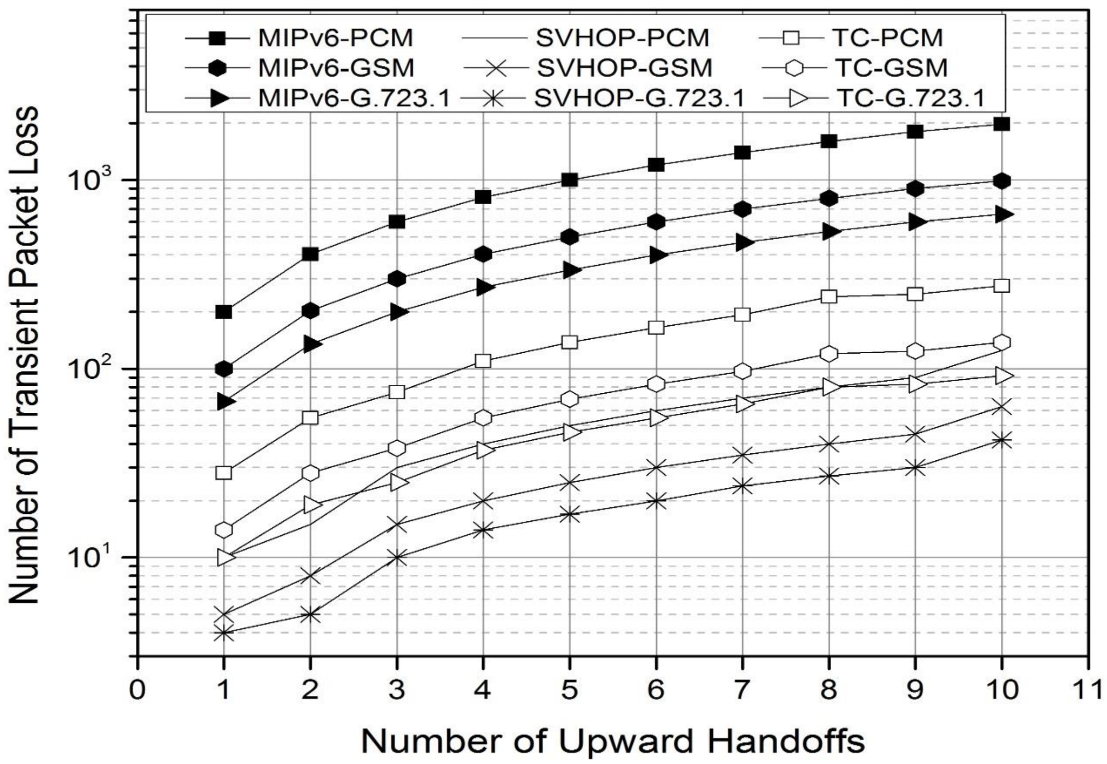

Figure 4.12: Number of transient packet loss during WLAN to UMTS handoff

implementation was to illustrate that different application can respond differently, even though the integration protocol remains the same.

Second, let us analyze the above graphs from the perspective of integration protocol. Since, the MIPv6 mechanism incurs highest session blackout time; therefore, it experiences highest transient packet loss. Conversely, the lowest session blackout time of SVHOP contributes to the least transient packet loss. The above graphs validate Equation 3.15 and 3.21 presented in the previous chapter.

Figures 4.13 and Figure 4.14 shows the transient lost information using different codec types during the first blackout time when the wireless client moves from the UMTS to WLAN and WLAN to UMTS network, respectively. Similar to the packet loss case, we can observe that the codec type highly influences on the transient lost information. According to Table 4.3, the PCM, GSM, G.726, G.729 and G.723.1 contains 80, 33, 40, 10 and 20 bytes in each packet, respectively. In general, it can be observed that the increase in packet size along with high packet arrival rate leads to the high transient lost information during the session blackout time. This observation is very significant in adopting the specific codec type into the scenarios where more or less number of handoff is somehow predictable.

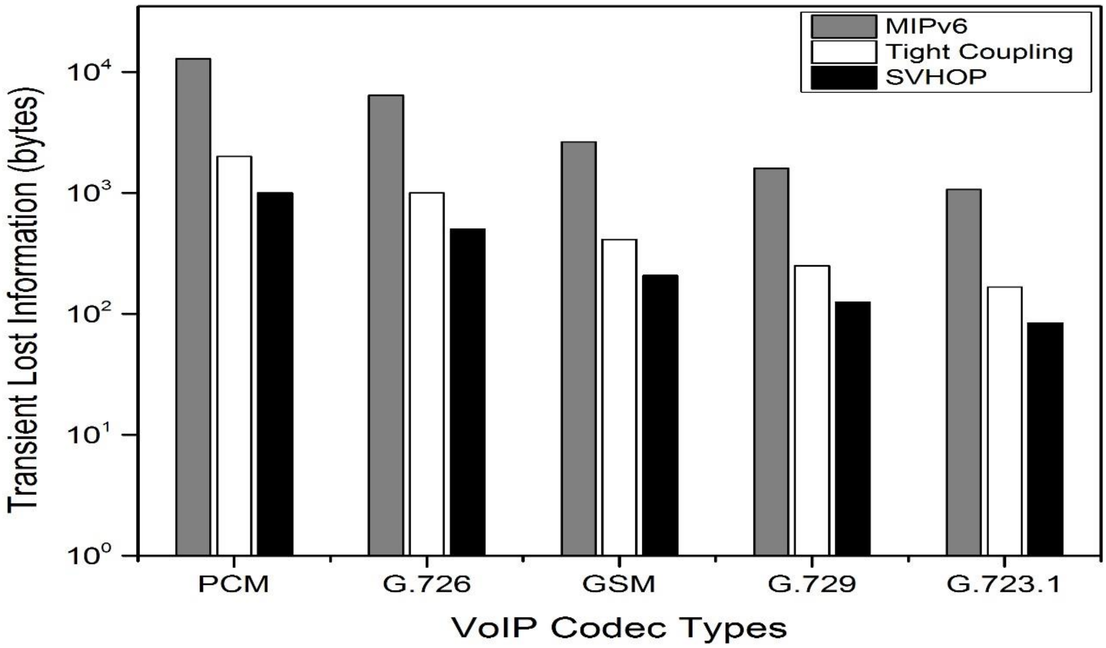

Figure 4.13: Transient lost information by using different codec types during UMTS to WLAN handoff latency

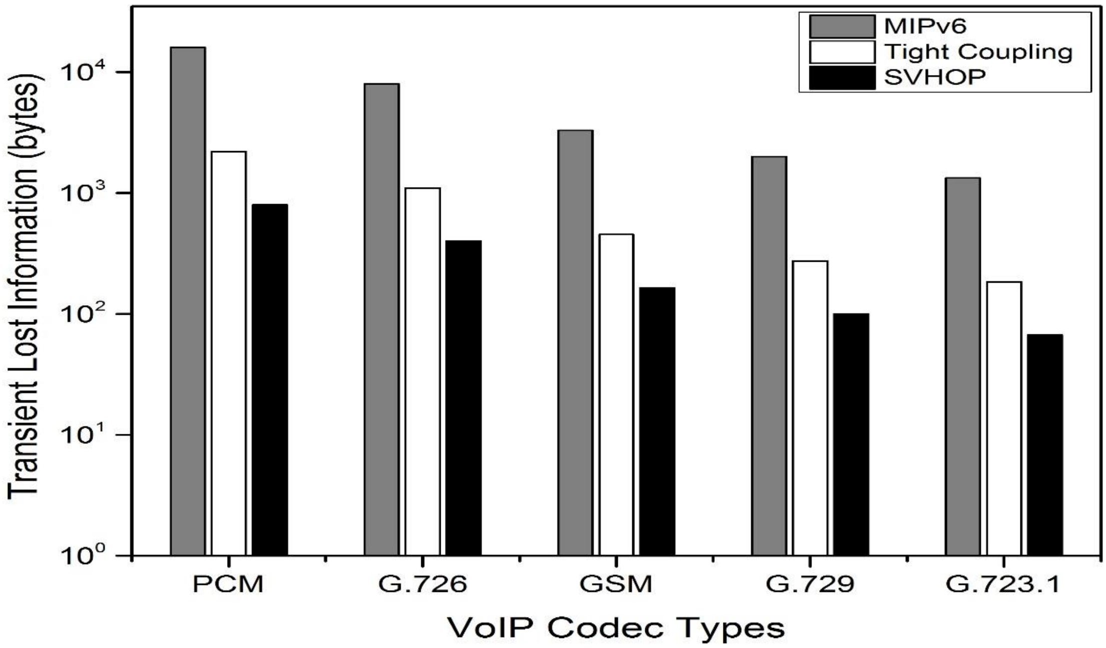

Figure 4.14: Transient lost information by using different codec types during WLAN to UMTS handoff latency

For example, in rural areas the UMTS cells are deployed with wider coverage. Moreover, fewer number of WLAN networks are expected. In such internetworking scenario when the wireless client moves from one point to another point, fewer handoffs are anticipated. Since, fewer numbers of handoffs produce less information lost; therefore, codec selection does not have a significant impact on the transient lost information.

On the contrary, in urban areas the UMTS cells are installed with the smaller coverage region. Therefore, to provide the coverage in a specific area more frequent UMTS cells are deployed. In addition, because of densely populated region high number of WLAN AP points are expected to be present. In such kind of internetworking scenarios, selection of smaller payload size codec type along with the low packet arrival rate will be a better option. Because more frequent handoff during a data session along with the high payload packet size codec will lead to the high amount of information lost.

It should be noted here that the PCM, G.726 and G.729 sends 100 packets per second. However, due to the different payload size, all these three codec types produce different lost information statistic. This observation cannot be measured by using the packet loss metric which does not include the payload size and, therefore, shows the identical lost when different traffic streams use similar packet arrival rate.

If we analyze the above graphs from the perspective of integration protocols, we can observe that the MIPv6 suffers with the highest information lost. On the other hand, the SVHOP provides the best performance compared to both benchmark protocols. The reason of this performance enhancement lies into the session blackout performance. High session blackout time leads to the higher lost information. Conversely, low session blackout time brings the lower lost information.

4.3.2 Stationary DMMT Scenario

Before sending the payload information over the wireless channel, there are several encapsulation overheads need to be appended by different layer of the sender device.

For example, the payload is encapsulated with the 12 bytes RTP header (for real time services), 8 bytes UDP header (for real time services), and 20 bytes TCP header (in case of non-real time data services) etc. Depending on the transmission medium, in case of WLAN, MAC layer introduces 34 bytes overhead. Nevertheless, if the UMTS RAN is used minimum 6 bytes additional bytes are introduced in the packet [91]. All these overheads are mandatory to append with the data packets. The sender device encapsulates the data packets, whereas, the receiving device decapsulates the payload by removing the overheads.

However, in between the sender and receiver several network devices and communication channels are present, which performance drops down because of additional information processing. The overheads influence the overall network performance in two ways. First, because of the high number of overheads, the end users and intermediate network nodes require longer processing time to process the packets. This processing delay increases the end to end delay and application response time. Second, since the bandwidth is a scarce resource, especially in a wireless environment, therefore, the process of overhead appending is a big obstacle to attain the high data service performance. By introducing the overhead information in the data packets, not only the data transmission but the overall network throughput is highly influenced. In several cases, the encapsulated packet contains more overheads compared with the original payload size. Therefore, in order to provide the optimal network performance, it is desired that any sophisticated integration protocol must avoid further adding its own overheads in the data traffic. As described in Section 4.2.1.2, in order to evaluate the influence of protocol overheads on the data service performance in an integrated UMTS/WLAN network, we consider the static DMMT located inside the WLAN access network and accessing the internet servers by using the MIPv6, tight coupling and SVHOP mechanism.

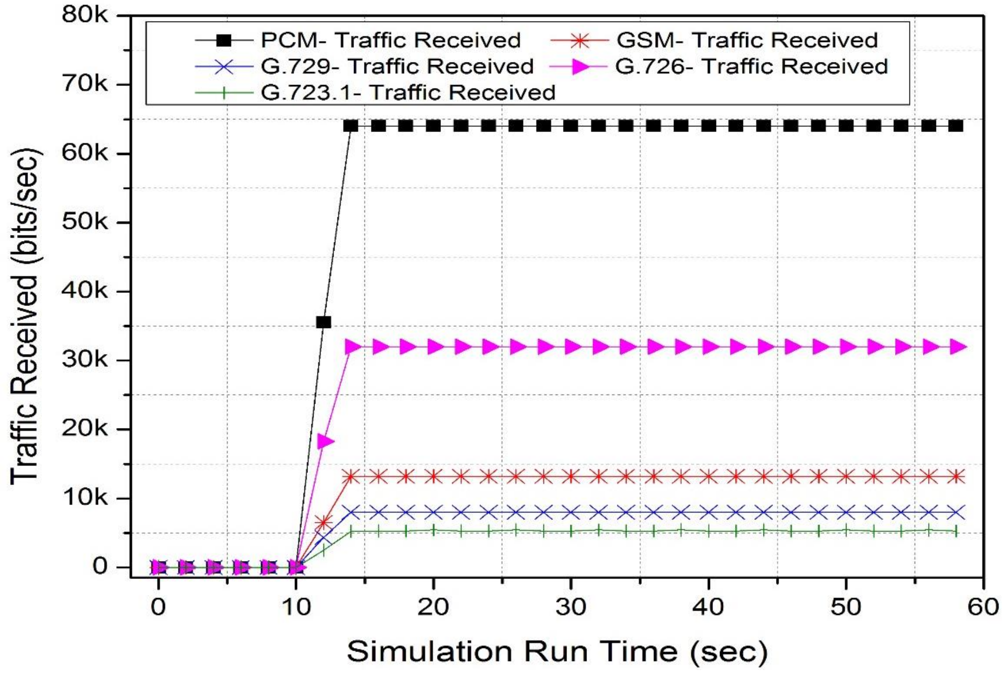

Figure 4.15: Tight coupling: Traffic received by using various VoIP codecs

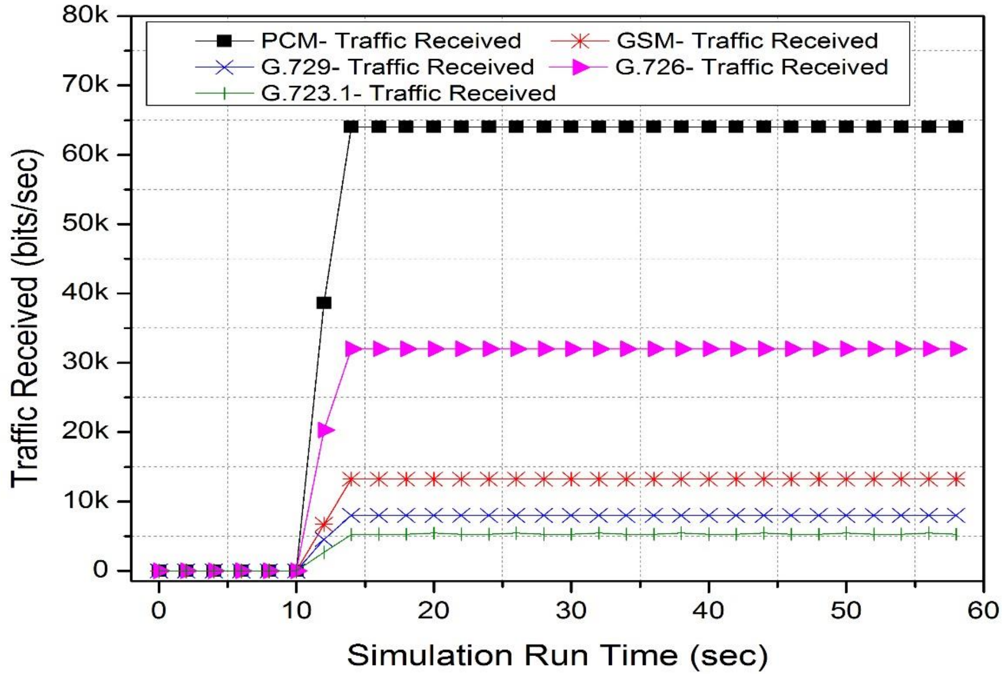

Figure 4.16: SVHOP: Traffic received by using various VoIP codecs

Figures 4.15 and Figure 4.16 illustrate payload traffic received by the DMMT in case of tight coupling and SVHOP mechanism, respectively. Five different variations of VoIP services have been analyzed that includes PCM, GSM, G.729, G.726, and G.723.1. The traffic received statistic shows that the DMMT receives 64 kbps, 13.2 kbps, 8kbps, 32kbps and approximately 5.3kbps when PCM, GSM, G.729, G.726, and G.723.1 encoders have been implemented, respectively. These simulation statistics satisfies the true default values of all the aforementioned codecs. It is worth mentioning here that in both tight coupling and SVHOP mechanism, no additional overhead belongs to the mobility management protocol is encapsulated by the sender. Consequently, the receiver only has to decapsulates the mandatory overheads such as RTP, UDP, IP, MAC, and physical layer headers etc.

In contrast with tight coupling and SVHOP mechanism, as reported by the literature, the MIPv6 extension headers are used by the CN/IS and the wireless device when the wireless client is located inside the foreign network. The extension headers are used to make the routing transparent to the upper layers and to avoid ingress

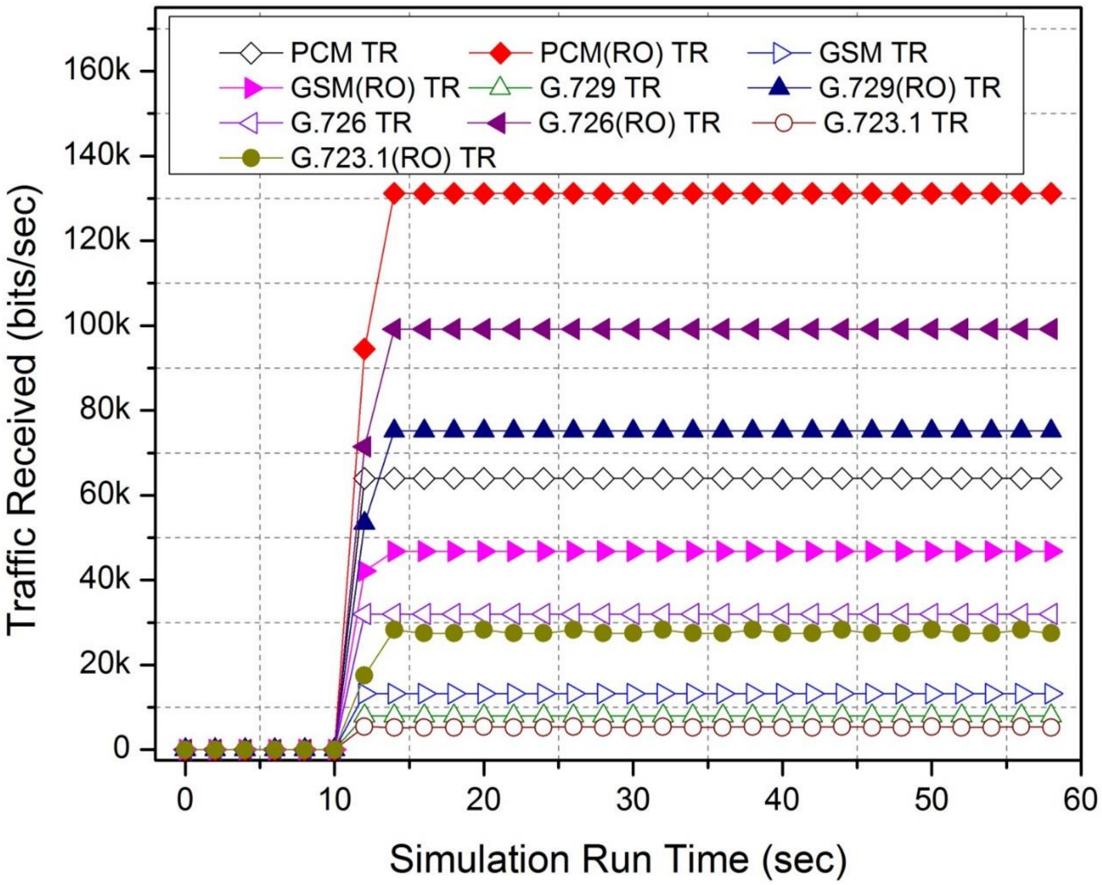

Figure 4.17: MIPv6: Traffic received and RO traffic received by using various VoIP codecs

filtering. The wireless client and IS encapsulates the data packets by using the home address option and type 2 routing header, respectively. The home address option with the destination extension header and type 2 routing header incurs 24 additional bytes each into the data packets for the communication [55, 79]. Figure 4.17 illustrates the data received information in case of MIPv6 mechanism. However, in addition, with the statistic belongs to the payload traffic received, the route optimized MIPv6 decapsulated data information from the MIPv6 module has also been collected and presented in Figure 4.17.

If we closely analysis the graph, we can see that the data traffic received, for example by using the PCM encoder, is 64kbps. However, the data decapsulated by the DMMT at the MIPv6 module is 131.2kbps. Similarly, the actual amount of data received by the DMMT is 13.2 kbps, 8kbps, 32kbps and approximately 5.3 kbps in case of GSM, G.729, G.726 and G.723.1 encoding schemes, respectively. Nevertheless, the corresponding DMMT decoding bits statistics at MIPv6 module shows that the received data is 46.8 kbps, 75.2 kbps, 99.2 kbps and 27.7 kbps. The main reason of these differences between the actual received packets by using different codecs and the corresponding decapsulated packets at the MIPv6 module is because of the additional 84 bytes overhead in every data packet. The formation of 84 additional bytes is as follows: 12 bytes RTP headers, 8 bytes UDP header, 40 bytes IPv6 header and 24 bytes MIPv6 extension headers. Appendix A and Appendix B demonstrates the MIPv6 Traffic Received (TR) and Route Optimized (RO) TR statistics and calculations, respectively.

According to the initial observations made on the collected statistics of Figures 4.15, 4.16, and 4.17 the conclusion drawn was in two folds. First, the MIPv6 must influence the network performance most compared to other analyzed protocols, since it introduces its own additional encapsulation overheads with the data packets. These overheads will not only degrades the core network entities' performance because of the high processing delay, moreover, the WLAN throughput will also be decreased. Second, both the intra-domain integration mechanisms i.e., tight coupling and SVHOP would affect the network performance much lesser than the MIPv6 mechanism. In addition, they degraded the network performance equally because both integration mechanisms only include the mandatory equal number of encapsulation overheads. Nevertheless, the latter part of conclusion was found to be partially true when the network data performance is evaluated by including more metrics.

Figure 4.18 and Figure 4.19 illustrates the average email and FTP download response time metric investigation, respectively, with the corresponding file sizes. The download response time represents the time elapsed between the requests send to the IS and receive the complete requested file from the IS. The connection setup signaling delay is also included in this time.

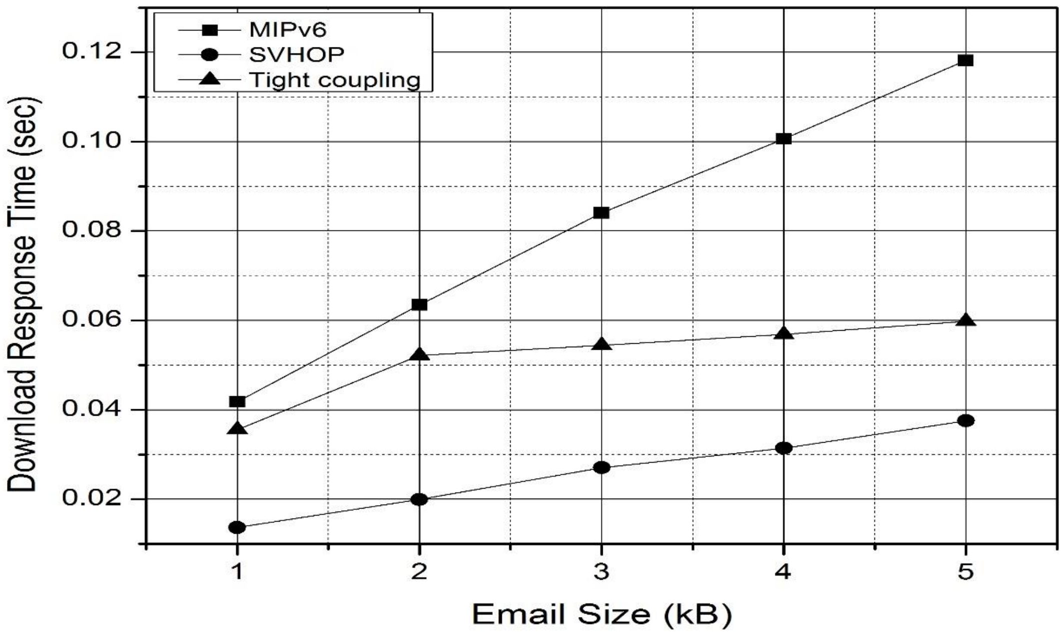

Figure 4.18: Email: average download response time

For diversity purposes, email and FTP file sizes of 1-5 and 100-1000 kilo-bytes, respectively, have been used for the course of three hours simulation run time. We can obverse the most rapid response by SVHOP mechanism compared with the MIPv6 and tight coupling mechanisms for both email and FTP services. For example, as illustrated in the Figure 4.18, for the case of 5 kilo-bytes email file, it takes about 0.038 seconds for the user to download the entire file by using the SVHOP. On the other hand, the same file can be downloaded in 0.118 and 0.060 seconds for the user accessing the IS by using the MIPv6 and tight coupling mechanisms, respectively. Similarly, as shown in the Figure 4.19, for the case of 1000 kilo-bytes FTP file, the

MIPv6 and tight coupling mechanism are required to download the complete file in 6.82 and 4.03 seconds, respectively. Nevertheless, the SVHOP downloads the same file in 2.67 seconds.

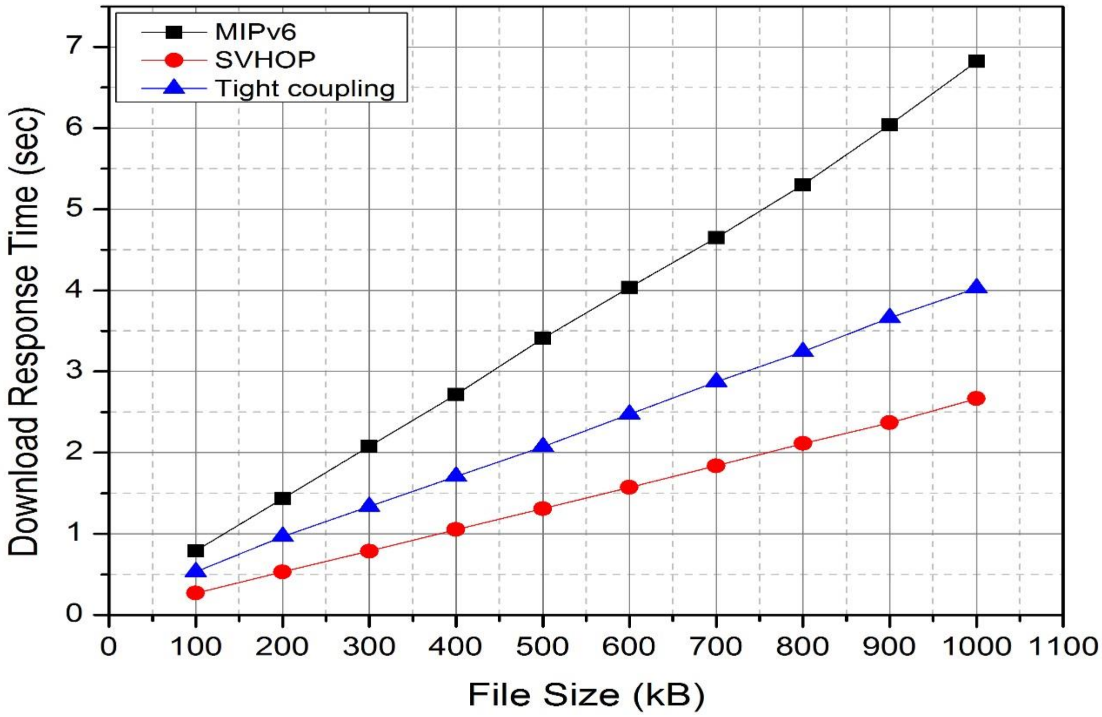

Figure 4.19: FTP: average download response time

Table 4.4 and Table 4.5 illustrate the average email and FTP upload response time metric investigation, respectively, with the corresponding file sizes. The upload response time represents the time elapsed between sending a file from DMMT to the IS and receiving the corresponding acknowledgement from the IS. This time also contains the connection setup signaling delay. Similar to the file download case, email and FTP file sizes of 1-5 and 100-1000 kilo-bytes, respectively, have been used in the course of three hours simulation run time. Similar to the download response time case, the SVHOP performs better than the other contemporary network integration techniques.

Table 4.4: Email: Average upload response time

Email File Size (kB)	Average Upload Response Time (sec)		
	MIPv6	Tight Coupling	SVHOP
1	0.043712	0.03756	0.015136
2	0.066394	0.054198	0.02005
3	0.11386	0.056411	0.025818
4	0.136864	0.059865	0.031985
5	0.15198	0.063978	0.037511

Table 4.5: FTP: Average upload response time

FTP File Size (kB)	Average Upload Response Time (sec)		
	MIPv6	Tight coupling	SVHOP
100	0.789237	0.49646	0.266907
200	1.435331	0.881569	0.533002
300	2.079073	1.273725	0.791831
400	2.720065	1.663443	1.051773
500	3.430762	2.054022	1.310311
600	4.005582	2.452289	1.572791
700	4.648186	2.847732	1.853225
800	5.326733	3.233202	2.106852
900	6.159762	3.633254	2.355796
1000	6.83569	4.038353	2.636594

For the web browsing, as shown in Figure 4.20, HTTP version 1.1 has been implemented for the simulation run time of three hours. The HTTP 1.1 addresses several issues of the 1.0 version that includes security, bandwidth utilization, connection management etc., [93]. Therefore, it is considered that the HTTP 1.1 is a true representative of current and future browsers. The HTTP page response time is

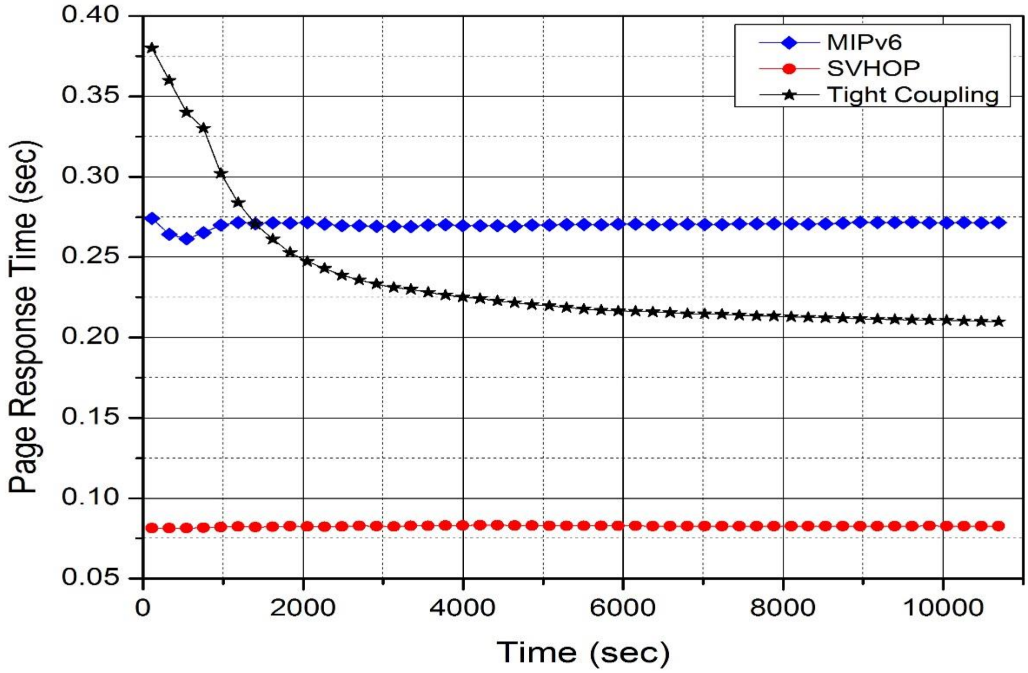

Figure 4.20: HTTP page response time

the time elapsed to retrieve an entire HTML page with all associated inline text and objects including the TCP connection setup time. It can be observed that the performance of the proposed SVHOP mechanism is dominant over MIPv6 and tight coupling mechanisms. The average page response time is respectively 0.083, 0.27 and 0.25 seconds for SVHOP, MIPv6 and tight coupling when they are implemented on the integrated UMTS/WLAN network. Since, in tight coupling and MIPv6mechanism downloading is slower than the proposed mechanism; therefore, the wireless client will have to wait longer to download the webpage. Hence, the internet surfing will be slower.

Figure 4.21 illustrates the video streaming end-to-end delay when all three protocols are implemented on the integrated UMTS/WLAN network. The end to end delay represents the time elapsed between packet is sent out by a video calling party to the time the packet reaches a video called party. It can be observed that the average end to end delay in case of MIPv6, tight coupling and SVHOP cases are 0.06211, 0.055459, and 0.0450, respectively.

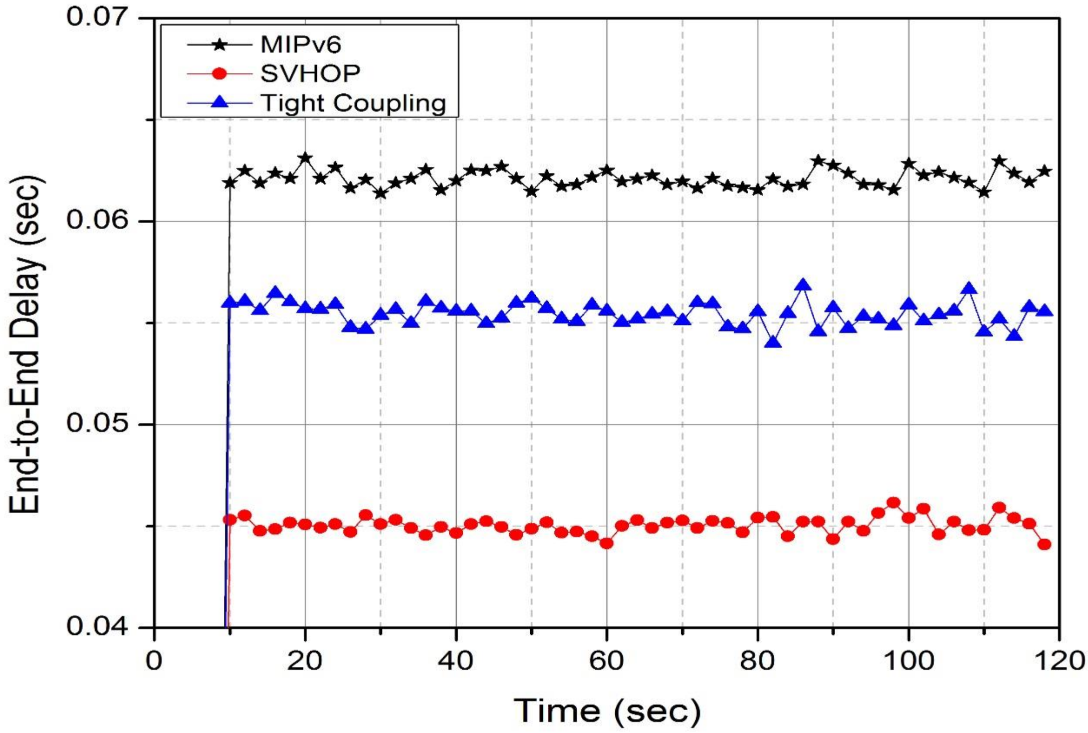

Figure 4.21: Video streaming: End to end delay

In order to further evaluate the data services' performance in an integrated UMTS/WLAN network the user density inside the foreign network was increased from 5 to 70 wireless users. For this simulation scenario, the FTP of file size 1000 kilo-bytes was used for the course of 30 minutes simulation run time.

Figure 4.22 plots the average aggregated/system traffic received (system throughput) metric investigation with the corresponding increasing number of wireless clients within the WLAN coverage region. Figure 4.22 highlights some very interesting points. First, similar to all previous cases where the traffic load has been increased gradually, the MIPv6 performance is visibly low compared to the SVHOP and tight coupling mechanism. Second, in case of MIPv6, it can be observed that the network performance reaches to its maximum throughput even when only 30 users are simultaneously requesting the FTP files from the internet server. Afterwards, as the number of wireless clients is increased, the overall network throughput gradually drops down. In contrast, in case of both tight coupling and SVHOP mechanism, the overall network throughput starts decreasing after 55 users simultaneously request the FTP server.

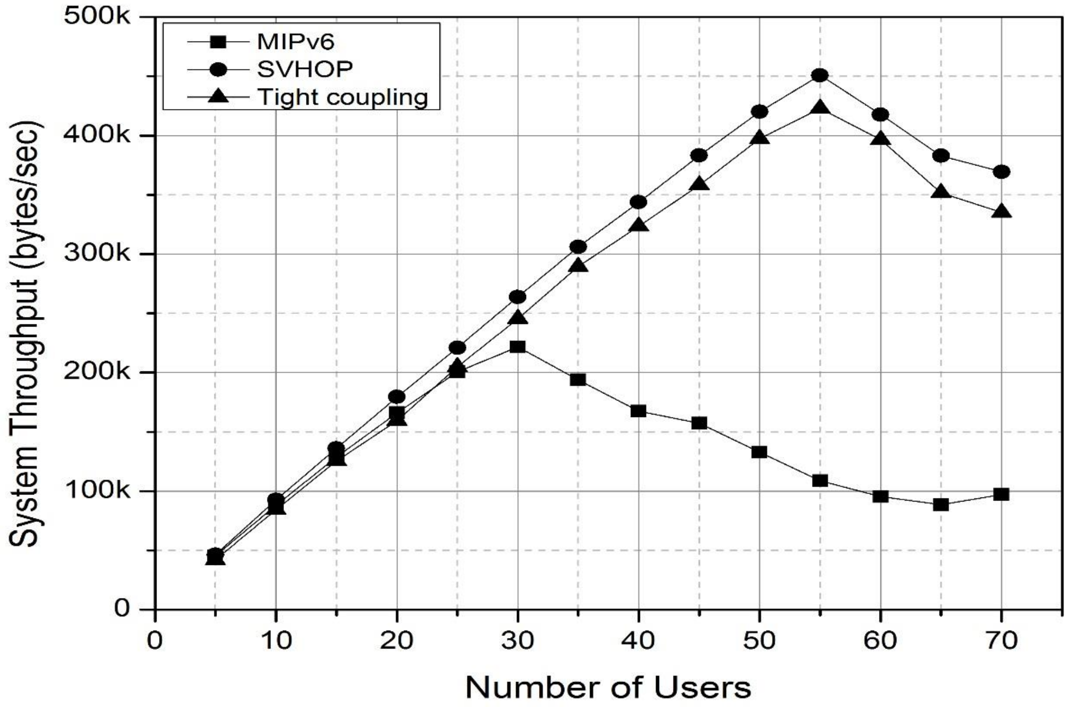

Figure 4.22: Effect on system throughput by varying number of network nodes

In general, the overhead consumes surplus wireless bandwidth as it requires additional time to transmit the data packets [94]. Therefore, the MIPv6 decreases the maximum throughput of the network because of its own high number of overhead in the data packets compared with the SVHOP and tight coupling mechanism in which no additional mobility management overheads are transmitted on the air interface. That is why, both intra-domain integration mechanism outperforms the MIPv6 mechanism. It must also be noted here, in case of SVHOP mechanism, the WLAN network throughput drops down when the system traffic received is 450 kilo-bytes which is equivalent to 3.6Mbps. The reason of the discrepancy between the connection rate and the maximum attained throughput is discussed below.

- As illustrated in Table 4.1, the WLAN 802.11b which provides the data rate of 11Mbps was used. It is worth mentioning here that the WLAN network maximum achievable throughput is significantly lesser compared to the actual connection rate. The maximum achievable WLAN throughput ranges lies much below than 50% [94-97] to approximately 60% [98] of the actual physical layer transmission rate. The relatively small throughput compared with the underlying physical layer transmission rate is attained because of several factors that include WLAN inter-frame space (IFS), MAC headers, PHY preamble/header, acknowledgement transmission etc. [95]. As reported in [97], over 50% of the total time is consumed in sending the MAC and PHY layer WLAN headers.

- There are several other elements that further reduce the WLAN throughput that includes the interference in the channel and increasing number of nodes [99]. The WLAN is a shared medium; therefore when there are multiple users simultaneously transmitting data on the radio channel then the wireless clients experience collision. Consequently, the transmitting nodes have to wait for a backoff time before attempting to retransmit. This results in lost airtime which affects the throughput of the system [100]. Consequently, when multiple users access the wireless network there is a huge variation in the maximum throughput [95, 96] that varies from 50% to 70% reduction in the maximum throughput [96].

At this stage it is worth referring back to the conclusion drawn on the basis of Figures 4.15, 4.16, and 4.17. The first part of the conclusion stated that the MIPv6 requires highest number of protocol overheads among all the analyzed protocols; therefore, it must affect the data performance most. This statement is also found true when the network performance was analyzed in terms of application response time and network throughput. The second part of the conclusion was SVHOP and tight coupling mechanism introduces an equal number of mandatory overheads. Therefore, the network performance should be influenced equally when either protocol is applied to the network. However, this part of the conclusion is now questionable when the tight coupling mechanism shows higher application response time and low throughput compared with the SVHOP protocol.

Figure 4.23 illustrates why the SVHOP performance is better than the tight coupling, even though the collected statistic showed the identical number of overheads.

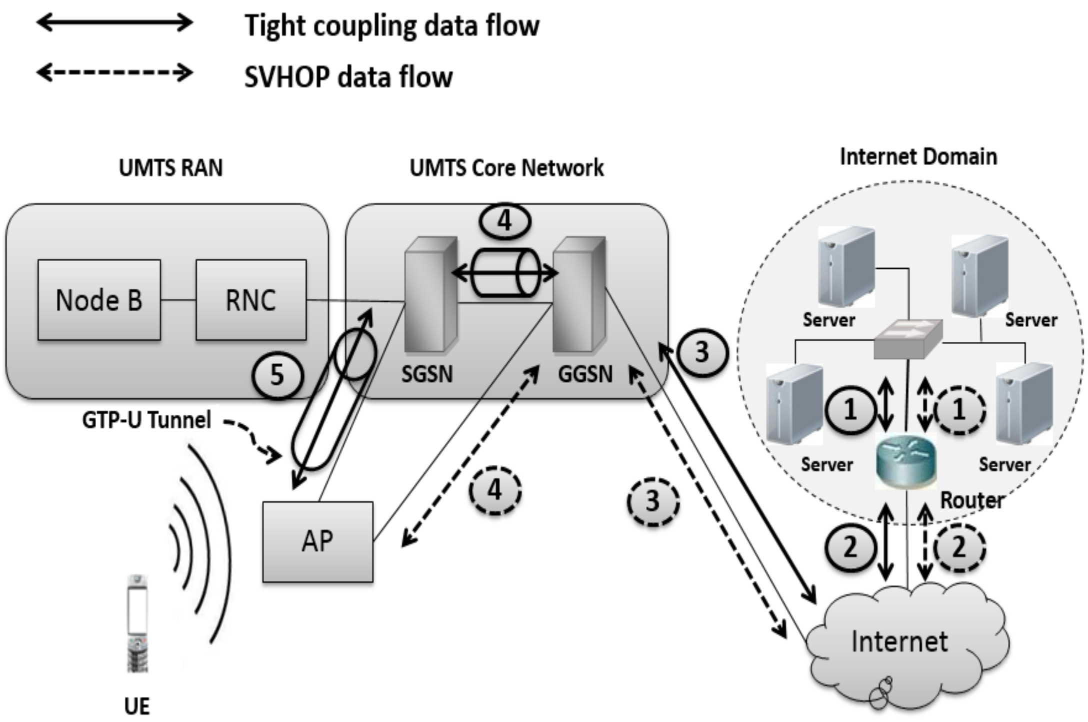

Figure 4.23: PDU routing from internet server to the wireless client

- As described in [101], it is desired that the PDUs should follow the optimal route to reach the destination. The optimal route is described as the route which requires the least number of network hops, moreover, alleviates the UMTS core network burden by bypassing the UMTS SGSN to reach to the destination. Figure 4.23 clearly illustrates that to send the PDUs from internet server to the DMMT; the SVHOP achieves the optimal route. In contrast to the tight coupling mechanism, in which PDU traffic follows: IS→ Router → Internet → GGSN → SGSN→ RNCE → DMMT, in SVHOP PDUs traverse IS → Router → Internet → GGSN → AP → DMMT. This means not only SGSN is bypassed; moreover, an optimal route is also attainted by decreasing number of network hops. Therefore, when the PDUs traverse with the longer path it takes higher time to download the complete file compared to the shorter route.

- According to the 3GPP technical specification 3GPP TS 29.060 [85], when the data is intended for the UMTS user from external data networks, the GTP-U tunneling is applied between the GGSN and SGSN over the Gn interface, moreover, between the SGSN and RNC over the Iu-PS interface. The wireless client connects to the UMTS core network without being aware of the GTP tunneling [80, 85]. The GTP tunnel introduces 12 bytes additional overheads on each data packet [80-83]. Since, in the tight coupling network integration mechanism, the intra-domain mobility management environment is provided by implementing the UMTS mobility management protocols over the integrated networks. Therefore, the GTP tunneling is present from the GGSN to RNC emulator for the data transportation. Nevertheless, the statistics collected over the DMMT in Figure 4.15 showed as there were no GTP tunneling overheads. Consequently, the network nodes such as RNC emulator, SGSN and GGSN require some additional processing time for every data packet to pass on. Conversely, in case of the proposed mechanism, the WLAN AP is connected with the GGSN with the G_i interface. As explained in previous chapter Section 3.3, here the GGSN appears as a simple router and sends the PDU without implementing any encapsulation on the air interface or at core network. This simplicity and straightforwardness of the proposed technique leads to the low latency and processing requirements for the data packets transportation compared to the tight coupling mechanism.

4.4 Protocol Evaluation and Comparison

In order to provide the ubiquitous connectivity and seamless mobility while the user is roaming across the integrated heterogeneous wireless network, several protocols have been proposed in the literature. The MIPv6 along with its route optimization mode of communication is the leading network layer integration solution up to date. On the link layer, the tight coupling integration mechanism has been widely suggested and implemented. In order to compare the proposed SVHOP integration protocol, MIPv6 with route optimization communication mode and tight coupling mechanism have been considered as the baseline protocols. Table 4.6 shows the comparison chart of the proposed and baseline simulated protocols. This comparison table summarizes the results collected in this chapter. It is quite clear that the proposed mechanism is an optimal choice compared with the baseline protocols for the integration of the NGNs.

Table 4.6: Summary of the simulated protocols

Metrics	SVHOP	MIPv6	Tight Coupling
Handoff delay	Low	High	Moderate
Session blackout time	Low	High	Moderate
Packet loss	Low	High	Moderate
Lost information	Low	High	Moderate
APOs	No	High	Low
Application response time	Fast	Very slow	Average
Network throughput	High	Low	High

4.5 Chapter Summary

This chapter comprehensively evaluated and analyzed the MIPv6, tight coupling and the proposed SVHOP protocols. By using the OPNET Modeler two different case studies were designed. The first simulation scenario evaluated the handoff delay, session blackout time, packet loss and lost information metrics while the wireless client was roaming across the UMTS and WLAN access networks. In the second simulation scenario, the additional protocol overheads, application response time, and network throughput of all the simulated protocols has been presented.

It was observed that the SVHOP provides optimum performance in terms of seamless mobility management by providing the low VHO delay and packet loss while the user is moving across the integrated access networks. From the simulation results, it is manifest that the SVHOP requires less VHO delay and packet loss for both upward and downward VHO cases compared with baseline protocols. This has been made possible by introducing the proactive RSS based VHO protocol, multi-homing feature, less signaling and node processing cost requirement in the proposed integration protocol.

When the DMMT is communicating with the internet servers via WLAN network, the SVHOP provides an optimal routing mechanism by decreasing the number of network hops to/from internet servers. Furthermore, unlike the MIPv6 and tight coupling mechanisms, no additional data encapsulation is required for the data transportation in the SVHOP. In consequence of the optimal routing technique and APO free mechanism, significant data service performance improvements have been achieved in terms of email and FTP download/upload response time, faster web browsing, and network throughput.

CHAPTER 5

CONCLUSION AND FUTURE WORKS

5.1 Chapter Overview

This chapter concludes the undertaken research. Moreover, illustrates the significance of the integrated wireless heterogeneous networks, key challenging issues of integrated network designs and the enhancements attained by implementing the proposed SVHOP mechanism as compared to the existing networks internetworking techniques. Finally, this chapter provides the recommendations for future work.

5.2 Research Conclusions

In order to provide the ubiquitous connectivity and seamless mobility while the user is roaming across the integrated heterogeneous wireless network, the IETF has proposed several mobility management protocols at different TCP/IP layers. Furthermore, the European Telecommunications Standards Institute (ETSI) has divided internetworking approach into two coupling mechanisms; Tight and loose coupling.

In this thesis, a comprehensive study has been conducted by reviewing several leading contemporary wireless integration protocols. It has been found that most of the loose coupling mechanism suffers from the high handoff latencies and packet losses. However, they do provide an independent integration mechanism for the existing networks without significant protocol modifications. Consequently, the loose coupling internetworking solutions lead to the ease of network deployment. In contrast to the loose coupling mechanisms, the tight coupling mechanism provides the best solution in terms of handoff latency and packet loss. Nevertheless, tight coupling is the most complex approach that can be implemented in an integrated network design as it requires major modification in the existing network protocols. Moreover, the tight coupling introduces additional network component emulators to establish compatibility among different networks. Since, the internet service providers/ telecommunication operators all over the world have already invested an enormous amount to the fully functional legacy networks; significant modifications in existing protocols, additional mobility management components or a global roll out with a new technology would not be easily accepted.

In this research, SVHOP has been introduced to bridge the gap between the tight and the loose coupling. More specifically, the SVHOP adopts the benefits, and at the same time avoids the shortcomings of loose and tight coupling approaches.

The key achievements of the SVHOP are:

- Seamless mobility
- Network implementing simplicity, i.e., ease of network implementation without any major modification to the existing network topology or protocol architecture and
- Enhance data performance in the foreign network.

The underlying principle of the SVHOP are based on the proactive RSS handoff triggering mechanism and multi-homing feature which reduces the signaling and processing cost of the network that are the main contributors of high vertical handoff delay and packet loss. Moreover, the proposed protocol is designed in a manner that it improves the data session performance because of its additional protocol overhead free mechanism and optimal route selection for data transportation in a foreign network of the integrated UMTS/WLAN network.

This dissertation comprehensively evaluated and analyzed the proposed SVHOP mechanism. In addition, extensive simulations are carried out to compare the performance of SVHOP with the base line contemporary approaches. By using the OPNET Modeler two different case studies were designed.

The first simulation scenario evaluated the handoff delay, session blackout time, packet loss and lost information metrics while the wireless client was roaming across the UMTS and WLAN access networks. In contrast with the base line protocols, the SVHOP mechanism is well aware of the handoff region by continuously monitoring the RSS information. Therefore, instead of establishing UMTS connectivity after losing the current session with the WLAN network, an early decision of handoff was executed. As a result of this proactive approach, the PDP context message is sent to UMTS network concurrently with the active data session with the WLAN network. Similarly, in case of UMTS to WLAN network switching, most of the session establishment messages are sent in parallel with the ongoing session. This consequently leads to low signaling and processing cost of the network and yield low handoff latency and packet loss during handoff. In addition with the handoff delay and packet loss, session blackout time and lost information metrics performance have been improved as compared to the MIPv6 and tight coupling mechanisms.

In the second simulation scenario, the additional protocol overheads and application response time of all the simulated protocols has been explored. For APOs investigation, several VoIP codecs such as PCM, GSM, G.726, G.729 and G.723.1 has been used. To further analyze the influence of the APO on the network application response time and network throughput different non-real time services such as FTP, Email and HTTP have been implemented. In contrast with the MIPv6 and tight coupling mechanism which appends 24 and 12 bytes of additional protocol overheads on every PDU, respectively, the SVHOP does not require any addition overheads. Moreover, when the DMMT is communicating with the internet servers via WLAN network, the SVHOP provides an optimal routing mechanism by decreasing the number of network hops to/from internet servers. In consequence of the optimal routing technique and additional protocol overhead free mechanism, the data service performance improvements have been achieved in terms of email and FTP download/upload response time, faster web browsing, and overall system throughput.

5.3 Research Contributions

This thesis makes significant contributions to the current state-of-the-art in the field of integrated wireless heterogeneous networks. The main contributions of this thesis are as follows:

- The most significant contribution of this research work is to propose, design, and evaluate the Seamless Vertical Handoff Protocol (SVHOP). To attain the seamless mobility, the foundations of the underlying principle of the SVHOP are based on the proactive RSS handoff triggering mechanism along with the optimized multi-homing capabilities of DMMT. By using these features, the SVHOP reduces the signaling and processing cost of the network that significantly contributes to achieve low handoff delay and low packet loss during upward and downward vertical handoff compared to the contemporary vertical handoff approaches. In addition, the proposed mechanism provides the ease of network deployment by developing "SIMPLE" integrated network architecture. The network design is simple as it does not require significant protocol alteration or introduction of additional network components in the existing UMTS and WLAN networks. Thus, for the mobility management, most of the UMTS and WLAN network components are reused without any modification.

- For the efficient transportation of data traffic inside the foreign network, an APO free mechanism has been introduced. As described earlier, most of the contemporary protocols such as MIP and tight coupling mechanism append high additional protocol overheads with the data packets. The APOs not only waste the scarce wireless bandwidth, moreover, slowdown the intermediate network nodes' speed by increasing their data processing time. The proposed APO free mechanism avoids the additional overheads for the data transportation by introducing an IP swapping mechanism. Consequently, it enhances the performance of data service by reducing the application response time of data services and by increasing the network throughput.

5.4 Future works

This section briefly discusses the future directions of the SVHOP mechanism for the integrated wireless heterogeneous networks. Following are research directions that can be explored to address the network integration issues more efficiently:

- For the proactive VHO decision mechanism, the RSS was considered as the key link layer metric. However, there could also be several other factors that may be taken into consideration when the handoff decision is made. These parameters can be velocity of the mobile terminal, current load of the network, the SNR of the channel etc. The points below explain why and in which scenarios velocity of mobile terminals, current load of the network, and SNR gains importance for the evaluation of the vertical handoff decision.

  a) In case of heterogeneous wireless networks, consider an urban area where multiple WLAN access points are overlaid by the UMTS cell. In this case, if the wireless client is moving with the high velocity, for example, the session is ongoing from the moving car then after every few seconds a vertical handoff would be triggered. Such vertical handoff can be categorized as the unnecessary handoffs. An increasing number of unnecessary vertical handoffs can lead to the increase in handoff blocking probability, packet loss and network throughput degradation etc. Therefore, to avoid such situations, the velocity of the mobile terminal should be taken into consideration.

  b) When performing a vertical handoff decision, if the actual load of the current and the target network is known then the best network among the multiple integrated networks can be selected for the ongoing session. For example, it might be the case that the DMMT suggest to perform a handoff to a network which is currently overloaded. As a result, when the handoff is executed to the overloaded network then the handoff session will be facilitated with very low network resources. Hence, the QoS of the ongoing session will be compromised. Furthermore, if the overloaded network is highly running out of network resources then the handoff request may also be rejected. Nevertheless, if the DMMT is aware with the current load of the target network before actually executing the handoff then smooth data session continuity can be assured.

c) In case of data sensitive application, especially in a noisy channel, the SNR can be considered as one of the best metric to measure the session Quality of Services (QoS). Therefore, when the communication is being performed in a noisy channel, an appropriate vertical handoff decision is expected if the vertical handoff includes the SNR.

To study the influence of the velocity, SNR, network load etc., on the vertical handoff decision, it requires further detailed literature study, hypothesis designing, analysis and evaluations, extensive simulations along with new vertical handoff techniques and protocols. Therefore, these issues can be explored in the future to address the network integration issues more efficiently.

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LIST OF PUBLICATIONS

Journal Publications

- **Safdar Rizvi**, and N.M. Saad, "*SSVHOP: A Simple and Seamless Vertical Handoff Protocol for Loosely Coupled Integrated UMTS/WLAN Networks*", Research Journal of Applied Sciences, Engineering and Technology. Volume 7, issue 6, pp. 1082-1092, February 2014. **(ISI and Scopus Indexed)**

- **Safdar Rizvi**, and N.M. Saad, "*A Proactive RSS based Vertical Handoff Protocol for Integrated UMTS/WLAN network*", Australian Journal of Basic and Applied Sciences, 2014. **(ISI and Scopus Indexed) (Accepted)**

- **Safdar Rizvi** and N.M. Saad, "*A Dual Mode Mobile Terminal for Enhanced Performance of Data services in NGN Networks*," EJSR. Volume 67, issue 3. pp. 360-377, Jan 2012. **(ISI and Scopus Indexed)**

- **Safdar Rizvi** , Asif Aziz, N.M. Saad, Nasrullah Armi and Mohd Zuki Yusoff , " *Tight Coupling Internetworking between UMTS and WLAN: Challenges, Design Architectures and Simulation Analysis* " , IJCN, Volume 3 , issue 2. pp. 116-134. 2011.

- **Safdar Rizvi** and N.M. Saad, "*A Proactive Cross Layer Vertical Handoff Approach for Seamless Mobility in Next Generation Networks*," The Scientific World Journal. **(ISI and Scopus Indexed) (Submitted)**

Conference Publications

- **Safdar Rizvi**, Asif Aziz, N.M Saad and Brahim Belhaouari Samir, "*Dual Mode Mobile Terminal for Integrated Tight Coupled UMTS and WLAN Network*", 2011 4th International Conference on Computer and Electrical Engineering (ICCEE 2011) Singapore. October 14-15, 2011.

- **Safdar Rizvi,** Asif Aziz and N.M. Saad,"*An Overview of Vertical Handoff Decision Policies for Next Generation Wireless Networks*", IEEE Asia Pacific Conference on Circuits and Systems, 6-9 Dec. 2010, Kuala Lumpur, Malaysia.

- **Safdar Rizvi,** Asif Aziz, N.M. Saad and Brahim Belhaouari Samir , "*A Comparative Analysis of Integration Schemes for UMTS and WLAN Networks*" , IEEE Asia Pacific Conference on Circuits and Systems, 6-9 Dec. 2010, Kuala Lumpur, Malaysia.

- **Safdar Rizvi** , Asif Aziz and N.M. Saad , "*Optimizations in Vertical Handoff Decision Algorithms for Real Time Services*" , 3rd International Conference on Intelligent and Advance Systems (ICIAS 2010), 15th- 17th June, 2010, Kuala Lumpur , Malaysia.

- **Safdar Rizvi,** N.M. Saad and Brahim Belhaouari Samir*," A comparative analysis of UMTS with the loosely coupled integrated UMTS/WLAN networks"*, 4rd International Conference on Intelligent and Advance Systems (ICIAS 2012), 12-14 June, Kuala Lumpur, Malaysia.

- **Safdar Rizvi** and N.M. Saad, "*A Multi-Homing Seamless Vertical Handoff Protocol for Integrated UMTS/WLAN Network*, 4rd International Conference on Intelligent and Advance Systems (ICIAS 2014), 3-5 June 2014, Kuala Lumpur , Malaysia. **(Accepted)**

<u>Other Publications</u>

- N.Armi, M.Arshad, **S.S.A.Rizvi**, M.Z.Yusoff, N.M.Saad "Performance Of Opportunistic Spectrum Access With Sensing Error In Cognitive Radio Ad Hoc Networks", Journal of Engineering Science and Technology (JESTEC), vol. 7, No. 2, pp. 142 – 155, April 2012. . **(Scopus Indexed)**

- Asif Aziz, **Safdar Rizvi** and N.M. Saad, "Fuzzy logic based vertical handover algorithm between LTE and WLAN," Intelligent and Advanced Systems (ICIAS), 2010 International Conference on, vol., no., pp.1-4, 15-17 June 2010. Malaysia.

- Asif Aziz, **Safdar Rizvi** and N.M. Saad, Samir, B.B., "An Overview of Integrated Architectures Solutions in Wireless Heterogeneous Networks," National Postgraduate Conference, (NPC 2011). Malaysia.

ACHIEVEMENTS

- Secured Bronze Medal at Science and Engineering Design Exhibition (SEDEX 32) , 11-12 December 2013.

APPENDIX A

MIPv6 TRAFFIC RECEIVED AND ROUTE OPTIMIZED TRAFFIC RECEIVED STATISTICS

MIPv6 PCM Statistics

time (sec)	TR (bytes/sec)	TR (pkts/sec)	RO TR (bits/sec)	RO TR (pkts/sec)
0	0	0	0	0
1	0	0	0	0
2	0	0	0	0
3	0	0	0	0
4	0	0	0	0
5	0	0	0	0
6	0	0	0	0
7	0	0	0	0
8	0	0	0	0
9	0	0	0	0
10	0	0	0	0
11	4320	54	0	0
12	8000	100	94464	72
13	8000	100	131200	100
14	8000	100	131200	100
15	8000	100	131200	100
16	8000	100	131200	100
17	8000	100	131200	100
18	8000	100	131200	100
19	8000	100	131200	100
20	8000	100	131200	100
21	8000	100	131200	100
22	8000	100	131200	100
23	8000	100	131200	100
24	8000	100	131200	100
25	8000	100	131200	100
26	8000	100	131200	100
27	8000	100	131200	100
28	8000	100	131200	100

29	8000	100	131200	100
30	8000	100	131200	100
31	8000	100	131200	100
32	8000	100	131200	100
33	8000	100	131200	100
34	8000	100	131200	100
35	8000	100	131200	100
36	8000	100	131200	100
37	8000	100	131200	100
38	8000	100	131200	100
39	8000	100	131200	100
40	8000	100	131200	100
41	8000	100	131200	100
42	8000	100	131200	100
43	8000	100	131200	100
44	8000	100	131200	100
45	8000	100	131200	100
46	8000	100	131200	100
47	8000	100	131200	100
48	8000	100	131200	100
49	8000	100	131200	100
50	8000	100	131200	100
51	8000	100	131200	100
52	8000	100	131200	100
53	8000	100	131200	100
54	8000	100	131200	100
55	8000	100	131200	100
56	8000	100	131200	100
57	8000	100	131200	100
58	8000	100	131200	100
59	8000	100	131200	100
60	8000	100	131200	100

MIPv6 G.729 Statistics

time (sec)	TR (bytes/sec)	TR (pkts/sec)	RO TR (bits/sec)	RO TR (pkts/sec)
0	0	0	0	0
1	0	0	0	0
2	0	0	0	0
3	0	0	0	0
4	0	0	0	0
5	0	0	0	0
6	0	0	0	0
7	0	0	0	0
8	0	0	0	0
9	0	0	0	0
10	0	0	0	0
11	530	53	0	0
12	1000	100	53392	71
13	1000	100	75200	100
14	1000	100	75200	100
15	1000	100	75200	100
16	1000	100	75200	100
17	1000	100	75200	100
18	1000	100	75200	100
19	1000	100	75200	100
20	1000	100	75200	100
21	1000	100	75200	100
22	1000	100	75200	100
23	1000	100	75200	100
24	1000	100	75200	100
25	1000	100	75200	100
26	1000	100	75200	100
27	1000	100	75200	100
28	1000	100	75200	100

29	1000	100	75200	100
30	1000	100	75200	100
31	1000	100	75200	100
32	1000	100	75200	100
33	1000	100	75200	100
34	1000	100	75200	100
35	1000	100	75200	100
36	1000	100	75200	100
37	1000	100	75200	100
38	1000	100	75200	100
39	1000	100	75200	100
40	1000	100	75200	100
41	1000	100	75200	100
42	1000	100	75200	100
43	1000	100	75200	100
44	1000	100	75200	100
45	1000	100	75200	100
46	1000	100	75200	100
47	1000	100	75200	100
48	1000	100	75200	100
49	1000	100	75200	100
50	1000	100	75200	100
51	1000	100	75200	100
52	1000	100	75200	100
53	1000	100	75200	100
54	1000	100	75200	100
55	1000	100	75200	100
56	1000	100	75200	100
57	1000	100	75200	100
58	1000	100	75200	100
59	1000	100	75200	100
60	1000	100	75200	100

MIPv6 GSM Statistics

time (sec)	TR (bytes/sec)	TR (pkts/sec)	RO TR (bits/sec)	RO TR (pkts/sec)
1	0	0	0	0
2	0	0	0	0
3	0	0	0	0
4	0	0	0	0
5	0	0	0	0
6	0	0	0	0
7	0	0	0	0
8	0	0	0	0
9	0	0	0	0
10	0	0	0	0
11	825	25	0	0
12	1650	50	42120	45
13	1650	50	46800	50
14	1650	50	46800	50
15	1650	50	46800	50
16	1650	50	46800	50
17	1650	50	46800	50
18	1650	50	46800	50
19	1650	50	46800	50
20	1650	50	46800	50
21	1650	50	46800	50
22	1650	50	46800	50
23	1650	50	46800	50
24	1650	50	46800	50
25	1650	50	46800	50
26	1650	50	46800	50
27	1650	50	46800	50
28	1650	50	46800	50
29	1650	50	46800	50

30	1650	50	46800	50
31	1650	50	46800	50
32	1650	50	46800	50
33	1650	50	46800	50
34	1650	50	46800	50
35	1650	50	46800	50
36	1650	50	46800	50
37	1650	50	46800	50
38	1650	50	46800	50
39	1650	50	46800	50
40	1650	50	46800	50
41	1650	50	46800	50
42	1650	50	46800	50
43	1650	50	46800	50
44	1650	50	46800	50
45	1650	50	46800	50
46	1650	50	46800	50
47	1650	50	46800	50
48	1650	50	46800	50
49	1650	50	46800	50
50	1650	50	46800	50
51	1650	50	46800	50
52	1650	50	46800	50
53	1650	50	46800	50
54	1650	50	46800	50
55	1650	50	46800	50
56	1650	50	46800	50
57	1650	50	46800	50
58	1650	50	46800	50
59	1650	50	46800	50
60	1650	50	46800	50

<u>MIPv6 G.726 Statistics</u>

time (sec)	TR (bytes/sec)	TR (pkts/sec)	RO TR (bits/sec)	RO TR (pkts/sec)
1	0	0	0	0
2	0	0	0	0
3	0	0	0	0
4	0	0	0	0
5	0	0	0	0
6	0	0	0	0
7	0	0	0	0
8	0	0	0	0
9	0	0	0	0
10	0	0	0	0
11	2160	54	0	0
12	4000	100	71424	72
13	4000	100	99200	100
14	4000	100	99200	100
15	4000	100	99200	100
16	4000	100	99200	100
17	4000	100	99200	100
18	4000	100	99200	100
19	4000	100	99200	100
20	4000	100	99200	100
21	4000	100	99200	100
22	4000	100	99200	100
23	4000	100	99200	100
24	4000	100	99200	100
25	4000	100	99200	100
26	4000	100	99200	100
27	4000	100	99200	100
28	4000	100	99200	100
29	4000	100	99200	100

30	4000	100	99200	100
31	4000	100	99200	100
32	4000	100	99200	100
33	4000	100	99200	100
34	4000	100	99200	100
35	4000	100	99200	100
36	4000	100	99200	100
37	4000	100	99200	100
38	4000	100	99200	100
39	4000	100	99200	100
40	4000	100	99200	100
41	4000	100	99200	100
42	4000	100	99200	100
43	4000	100	99200	100
44	4000	100	99200	100
45	4000	100	99200	100
46	4000	100	99200	100
47	4000	100	99200	100
48	4000	100	99200	100
49	4000	100	99200	100
50	4000	100	99200	100
51	4000	100	99200	100
52	4000	100	99200	100
53	4000	100	99200	100
54	4000	100	99200	100
55	4000	100	99200	100
56	4000	100	99200	100
57	4000	100	99200	100
58	4000	100	99200	100
59	4000	100	99200	100
60	4000	100	99200	100

<u>MIPv6 G.723.1 Statistics</u>

time (sec)	TR (bytes/sec)	TR (pkts/sec)	RO TR (bits/sec)	RO TR (pkts/sec)
1	0	0	0	0
2	0	0	0	0
3	0	0	0	0
4	0	0	0	0
5	0	0	0	0
6	0	0	0	0
7	0	0	0	0
8	0	0	0	0
9	0	0	0	0
10	0	0	0	0
11	300	15	0	0
12	680	34	17472	21
13	660	33	28288	34
14	680	34	27456	33
15	660	33	27456	33
16	660	33	28288	34
17	680	34	27456	33
18	660	33	27456	33
19	660	33	28288	34
20	680	34	27456	33
21	660	33	27456	33
22	660	33	28288	34
23	680	34	27456	33
24	660	33	27456	33
25	660	33	28288	34
26	680	34	27456	33
27	660	33	27456	33
28	660	33	28288	34
29	680	34	27456	33

30	660	33	27456	33
31	660	33	28288	34
32	680	34	27456	33
33	660	33	27456	33
34	660	33	28288	34
35	680	34	27456	33
36	660	33	27456	33
37	660	33	28288	34
38	680	34	27456	33
39	660	33	27456	33
40	660	33	28288	34
41	680	34	27456	33
42	660	33	27456	33
43	660	33	28288	34
44	680	34	27456	33
45	660	33	27456	33
46	660	33	28288	34
47	680	34	27456	33
48	660	33	27456	33
49	660	33	28288	34
50	680	34	27456	33
51	660	33	27456	33
52	660	33	28288	34
53	680	34	27456	33
54	660	33	27456	33
55	660	33	28288	34
56	680	34	27456	33
57	660	33	27456	33
58	660	33	28288	34
59	680	34	27456	33
60	660	33	27456	33

APPENDIX B

MIPv6 TRAFFIC RECEIVED AND ROUTE OPTIMIZED TRAFFIC RECEIVED CALCULATIONS

PCM (G.711)

Traffic Received
64,000 bits/sec
100 packets/second
Therefore, the payload size will be
= 64000 (bits/sec)/100 (packets/sec)
→ 640 bits/packet
→ 80 bytes/packet

RO Traffic Received
131,200 bits/sec
100 packets/second
Therefore, the payload size will be
= 131200 (bits/sec)/100 (packets/sec)
→ 1312 bits/packet
→ 164 bytes/packet

Encapsulation Overheads
164 bytes/packet - 80 bytes/packet = 84 bytes/packet pure overheads
IPv6 header = 40
UDP header = 8 bytes
RTP header = 12 bytes
MIPv6 extension header = 24 bytes

G.729

Traffic Received
8,000 bits/sec
100 packets/second
Therefore, the payload size will be
= 8000 (bits/sec)/100 (packets/sec)
→ 80 bits/packet
→ 10 bytes/packet

RO Traffic Received
75,200 bits/sec
100 packets/second
Therefore, the payload size will be
= 75200 (bits/sec)/100 (packets/sec)
→ 752 bits/packet
→ 94 bytes/packet

Encapsulation Overheads
94 bytes/packet - 10 bytes/packet = 84 bytes/packet pure overheads
IPv6 header = 40
UDP header = 8 bytes
RTP header = 12 bytes
MIPv6 extension header = 24 bytes

G.726

Traffic Received
32,000 bits/sec
100 packets/second
Therefore, the payload size will be
= 32000 (bits/sec)/100 (packets/sec)
→ 320 bits/packet
→ 40 bytes/packet

RO Traffic Received
99,200 bits/sec
100 packets/second
Therefore, the payload size will be
= 99200 (bits/sec)/100 (packets/sec)
→ 992 bits/packet
→ 124 bytes/packet

Encapsulation Overheads
124 bytes/packet - 40 bytes/packet = 84 bytes/packet pure overheads
IPv6 header = 40
UDP header = 8 bytes
RTP header = 12 bytes
MIPv6 extension header = 24 bytes

GSM

Traffic Received
13,200 bits/sec
50 packets/second
Therefore, the payload size will be
= 13200 (bits/sec)/ 50 (packets/sec)
→ 264 bits/packet
→ 33 bytes/packet

RO Traffic Received
46,800 bits/sec
50 packets/second
Therefore, the payload size will be
= 46800 (bits/sec)/ 50 (packets/sec)
→ 936 bits/packet
→ 117 bytes/packet

Encapsulation Overheads
117 bytes/packet - 33 bytes/packet = 84 bytes/packet pure overheads
IPv6 header = 40
UDP header = 8 bytes
RTP header = 12 bytes
MIPv6 extension header = 24 bytes

G.723.1

Traffic Received
5,334.4 bits/sec
33.34 packets/second
Therefore, the payload size will be
= 5,334.4 (bits/sec)/ 33.34 (packets/sec)
→ 160 bits/packet
→ 20 bytes/packet

RO Traffic Received
27722.24 bits/sec
33.32 packets/second
Therefore, the payload size will be
= 27722.24 (bits/sec)/ 33.32 (packets/sec)
→ 832 bits/packet
→ 104 bytes/packet

Encapsulation Overheads
104 bytes/packet - 20 bytes/packet = 84 bytes/packet pure overheads
IPv6 header = 40
UDP header = 8 bytes
RTP header = 12 bytes
MIPv6 extension header = 24 bytes

APPENDIX C

GTP HEADER FIELDS AND THEIR EXPLAINATION

As shown in Figure 3.5 in chapter 3, the GTP header consists of the following fields:

- Version field: The version field refers the GTP version being used. For the 2.5G network the set value is ‘0’, whereas, ‘1’ is set for the 3G network.
- Protocol Type (PT): This field contains the value ‘1’ or ‘0’. One refers to GTP and zero refers to GTP’. The GTP’ is used for the transportation of charging information and implements between the UMTS Charging Data Function (CDF) to the Charging Gateway Function (CGF).
- Extension Header flag (E): The presence of the Next Extension Header field is indicated by the E flag. When the E flag is set to ‘0’, the Next Extension Header is not present in the GTP header. When the E flag is set to ‘1’, it shows that the Next Extension Header is present in the GTP header.
- Sequence Number flag (S): The presence of the Sequence Number field is indicated by the S flag. When the S flag is set to ‘0’, the Sequence Number field is not present in the GTP header. When the S flag is set to ‘1’, it shows that the Sequence Number field is present in the GTP header.
- N-PDU Number flag (PN): The presence of the N-PDU Number field is indicated by the PN flag. When PN flag is set to ‘0’, the N-PDU Number field is not present in the GTP header. When the PN flag is set to ‘1’, it shows that the N-PDU Number field is present in the GTP header.
- Message Type: The message type field specifies the type of GTP message used. Since, this field contains 8 bits; therefore, 255 different values indicate various GTP-c and GTP-u types. All 255 variations of GTP messages are listed in the UMTS specification.
- Length: The length of the payload is identified by the Length field. The optional fields of the GTP header such as, Sequence Number, the N-PDU

Number or any Extension Headers are also considered as the part of the payload.

- Tunnel Endpoint Identifier (TEID): The Tunnel Endpoint Identifier is used to unambiguously identify the tunneling endpoint. The TEID values are exchanged between the UMTS nodes with the help of the GTP-c protocol at the Gn interface and by using the RANAP protocol on the Iu interface.

- Sequence Number: In the GTP-c, the Sequence Number field is used as a transaction identity for signaling. The response message header copies this value from the request message header. For GTP-u, this field is used to preserve the sequence order of the data.

- N-PDU Number: The N-PDU Number field is used when the wireless client performs inter SGSN Routing Area (RA) update and inter-system handoff procedures such as between 3G and 2G access networks.

- Next Extension Header Type: The Next Extension Header Type is used to indicate the type of extension header used in the GTP payload. PDCP PDU number, suspend request and suspend response are three different types of extension headers used in the GTP.